GT8 User Manual

Release 2.3

Welcome

Welcome to the GT8 VoIP Gateway User manual. This document covers features, functionalities, and installation procedures for the GT8 series, and provides tested configuration examples for our gateway users. After reading this book, you will learn more about the gateway, get familiar with the installation process, and feel more comfortable in using the software to perform all administrative activities.

Version:

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1

PRODUCT INTRODUCTION

Overview

GT8 is a multi-purpose VoIP gateway product series designed with the needs of service providers and enterprises in mind. With GT8 gateways, service providers can provide telephony and fax services to subscribers using many access methods such as FTTB, HFC, and ADSL. Enterprises can use the GT8's traditional PBX interface to implement voice VPN solutions with their private IP or public VPN networks. GT8 can also serve as a remote SIP terminal for IP-PBX solution.

GT8 has a variety of models. Each model can be customized to have different number of FXS ports and FXO ports. It shares the same software system as other GED's VoIP products (GT48 and GTT) and therefore keeps the advantages in functionality, quality, and compatibility of GED products. In hardware GT8 uses Motorola's MPC852 as the Central Processing Unit, and TI C5509 high efficiency chip to process voice and faxes. The powerful hardware equipment ensures GT8 to send signaling and IP packets in different channels even when traffic is at the peak, thus supports major functions such as voice codec (G.711, G.729A, G.723.1, GSM and iLBC) and echo cancellation.

This manual is mainly about GT8 installation and web configuration. Please note that after you have made changes to many of the parameters on GT8 Web Configuration page and clicked the Submit button, you may get messages like "Submission is successful. Please

restart the system to make the changes effective." You need to restart GT8 using the instruction in Restart gateway.

GT8 also has the capability to restore the default settings. Just click the Restore Default button.

GT8 configuration parameters have brief descriptions. To find out, just point your mouse over the parameter.

Features

GT8 has the following features:

- It supports SIP/MGCP protocols
- It supports route selection (it can route a call or direct it to the internet according to the called number)
- It supports RADIUS based CDR protocol
- It supports gain adjustment to FXS/FXO ports
- It supports the intrusion into NAT through a STUN server
- It supports traditional terminal devices, including phones, fax, and PBX
- It supports a variety of supplementary services such as All forward, Forward No Answer, Forward Busy Line, Call waiting, and Distinctive Ring, etc.
- It can obtain static IP address or capture mobile IP address through DHCP and PPPoE
- It supports the traditional fax service using T.30 and T.38 formats
- GT8 with FXO ports
- It supports the following sinaling protocols:
    - SIP (Compliant to RFC 3261 and TISPAN)
    - MGCP
- It supports the following codec:
    - G.711
    - G.723.1
    - G.729A
    - GSM

- iLBC
- G.168 Echo Cancellation
- DTMF RFC2833 and T.38

Hardware Platform

Physical

Figure 0-1 GT8 Front View

①	Reset button (RST)
②	Power indicator (PWR). If lit, power is on
③	Ethernet port indicator. If lit, it is in operation
④	FXS/FXO indicators. The port number is lit when in use

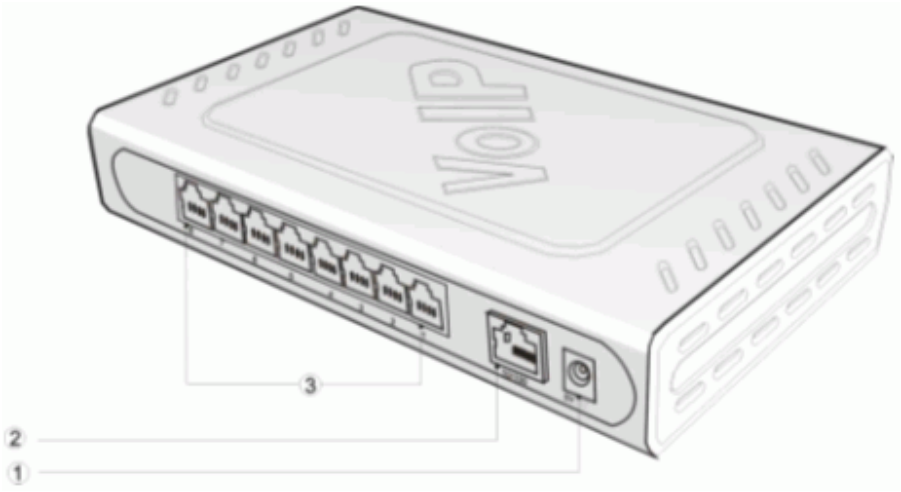


Figure 0-2 GT8 Rear View

①	Power plug-in
②	10/100 baseT Ethernet port
③	FXS/FXO ports, a total of 8

Table 0-1 GT8 Configuration Options

Model Number	RJ11 Port Configuration							
	1	2	3	4	5	6	7	8
GT8-4S	FXS1	FXS2	FXS3	FXS4	Null	Null	Null	Null
GT8-8S	FXS1	FXS1	FXS3	FXS4	FXS5	FXS6	FXS7	FXS8
GT8-/4	FXO1	FXO2	FXO3	FXO4	Null	Null	Null	Null
GT8-4S/4	FXS1	FXS2	FXS3	FXS4	FXO1	FXO2	FXO3	FXO4

System Specifications

Table 0-2 GT8 Specification

	Specification
Internal Memory	32MB
Flash Memory	4MB
On-hook Battery	-56V
Off-hook Battery	-24V
Ringing Voltage	60V
REN Equivalence	5 for short loop (1000 feet), 3 for long loop (5000 feet)
Loop Current	= or > 21 mA
Loop Resistance	Up to 188 Ω
Surge Voltage	Level two surge protection. Can stand up to 1000V (10/100uS) power surge
Max Line Length	1500 m
Off-hook Detection	Loop Start
Dialing	DTMF
Input Voltage	12V DC
Input Current	1.5Amp (Max)
Power Consumption	15Watt (Max)
Operation Temperature	0 ~ 40°C
Non Operation Temperature	–25 ~ 70°C
Operation Humidity	5 ~ 95% (Non Condensed)
Dimension (H×L×W)	300x190x45 mm
Weight	800g

2

PREPARATION FOR INSTALLATION

To avoid any body injury and device damage, please read this chapter carefully before the installation.

Safety Check

Please follow the safety guidelines when installing GT8.

- Keep away from wet group and heat
- Ensure safe use of electricity
- Ensure to connect all the interface cables correctly

Installation Environment

Temperature/Humidity

The GT8 installation room must maintain normal temperature and humidity.

If the room temperature exceeds the specified maximum temperature, it will shorten the live of the electrical insulation material. If the room humidity exceeds the specified humidity, GT8 may experience electrical static shock and shrinkage of electric insulation material in the metal package. It may also cause metal corrosion. All these will drastically shorten the life span of the GT8. It is strongly recommended that user control the environmental temperature

between **0 ~ 40ºC** and humidity between **5% ~ 95%** (**none condensing**).

Dust Control and Air Flow

Dust falls on the GT8 might cause intermittent failure in electrical connections. It may cause long term damage to GT8 will cause equipment failure and shorten equipment life span. Therefore, GT8 needs to have ample air flow in front of the GT8 air intake and outtake for proper heat exhaust.

Interference and Lightening Hazard

GT8 may experience various types of EMI hazards in operation and its performance may be impacted. To reduce those hazards, it is suggested that：

- Do not install GT8 close to high power wireless equipment, RADAR transmission site, and high frequency high electric current devices.
- GT8 comes with Level 2 lightening protection. Its operation site requires Level 1 lightening protection.
- GT8 must have its own power source and should be electrical interference free
- Ensure proper grounding

Installing GT8

When installing the GT8 please make sure GT8 is secured and has ample space for air flow.

Inspecting GT8 and Its Accessories

After the installation preparation is completed, the shipping package can be opened to examine all the items in the package. The list of items for the GT8 is shown in Table 2 - 1.

Table 0-1 GT8 Basic Configuration and Accessories

Model Number	Qty	Description
GT8-4S,GT8-8S, GT8-/4, GT8-4S/4	1	Each GT8 may have 4 FXS ports, or 8 FXS ports, or 4 FXO ports, or 4 FXS/FXO ports. You need to examine carefully to make sure what you receive is what you paid for
MX-PWR10-V01-00	1	GT8 DC adaptor 12V 1.5A
MX-CBL00-0005	1	5 meter Ethernet cable, 1.5m in length
MX-CBL00-0011	1	GT8 power cord

Note： It is suggested that users carefully examine the content of the shipping package according to the sales contract. If there is any question or problem, please contact our customer service department.

3

INSTALLATION

Installing GT8

Since GT8 is small, you can put it to a clean and flat workspace. Please make sure it is secured and has ample space for air flow.

Connecting the Cables

Connecting the Ethernet Port

GT8 has one 10/100 Base-T Ethernet port with RJ45 connector. It is equipped with LED status display. Besides voice packet, this port can also manage, maintain, and control the information flow.

The Ethernet Cable needs to be carefully made to ensure IP data and voice quality. The following is the Ethernet cable making scheme:

Step1： A user can use a proper cable peeling cutter to peel away 3cm skin of a CAT-5 cable. What is left is shown in Figure 3-1.

Figure 0-1

Step2： Twisted pairs. Currently, the most commonly used standard wiring scheme is EIA/TIA T568B shown in Figure 3-2. In the wiring scheme, pin 1 and 2 are a pair, pin 3 and 6 are a pair, pin 4 and 5 are a pair and pin 7 and 8 are a pair. According to the Figure 3-2, twisted pairs line up with colors (1: white orange，2: orange，3: white green，4:blue，5: white blue，6:green，7: white brown，8: brown). It is specially noted that the green and white green are separated by a pair of blue wires. It is a common mistake to put green and white green close together, which will result in interference and therefore lower transmission efficiency.

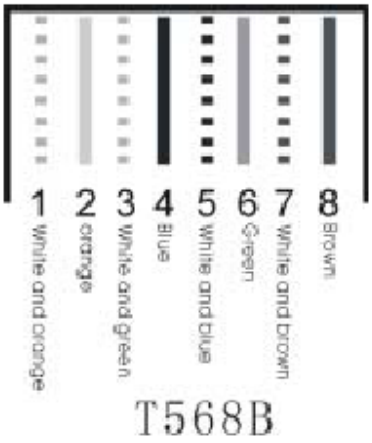


Figure 0-2 T568B wire pairing scheme

Step3： After lining up wires to the correct pin positions, trim all the twisted pairs with a cable cutter, leaving 15mm leads exposed. Then follow Figure 3-3 by inserting wires to their corresponding pin position in the plastic shell of RJ45 connector. Pin 1 will house white orange wire, etc.

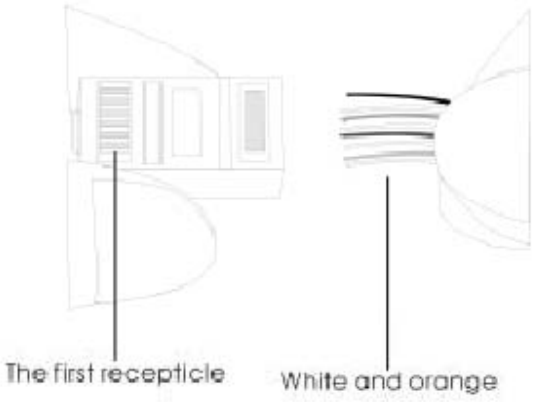


Figure 0-3 RJ 45 Wiring

Step4： After wires have been properly inserted into RJ45 connector; a cramping tool can secure the wires to the connector and make connections to the metal pins as shown in Figure 3-4.

Figure 0-4 Finished RJ 45

Since this is a direct connection, the connector for the other end of the cable can be made the same way using RJ45 connector.

After the Ethernet cable is ready, Connect one end of the cable to GT8's WAN port and the other end to a switch or router. Check the Ethernet status display: light or flash means activity.

Connecting FXS Cable

GT8 have FXS ports that connect to phones.

Connect one end of the RJ11 cable to the GT8 FXS port, and connect the other end to a phone, fax, or PBX.

Connecting FXO Cable

Certain GT8 products, like GT8-4 or GT8-4S/4, have FXO ports that connect to PBX or PSTN.

Connect one end of the RJ11 cable to the GT8 FXO port, and connect the other end to a PBX or PSTN line.

Connecting the Power Supply

Before plugging GT8 into the power outlet, it is suggested that tri-phase power outlet be used and grounding be properly connected.

Please follow the procedure when connecting to the power source：

Step1： Plug the DC head of the power adaptor into GT8's DC input socket.

Step2： Plug the AC head of the power adaptor into the power outlet of 110V or 220V.

Step3： Check to see if the PWR LED indicator is lit. If PWR LED is lit, everything is normal. If not, repeat Steps 1 to 2.

Note: If power up fails repeatedly, please contact GED technical support. Do not attempt to open GT8 to fix any problems.

Final Checks after Installation

After installing GT8 and before it is powered on, please make sure of the following:

- There is ample air space around GT8 for heat exhaustion.
- Power cord is standard and matches the required electric voltage.
- Make sure the ports are connected to the right devices.

Note：It is very important to recheck all the installation work to ensure GT8 to function properly and trouble free.

4

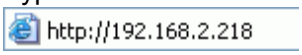
FUNCTION DESCRIPTION

Registration

Step1： Power up the GT8. GT8 by default uses DHCP, and will automatically detect an IP address; if you cannot get the IP address (when you connect to the computer directly), use default IP address “192.168.2.218”. After power up (when customer line LCD stops flashing), if the gateway uses MGCP protocol，it will tell the IP address to any first off-hook user; if using SIP protocol, you can press “##” to get the IP address through any customer line at any time.

Step2： Double click to open IE Explorer in the computer which is connected to the same network as GT8.

Step3： Type in GT8 IP address（for example：192.168.2.218）



，and the web interface will display as shown in Figure 4-1.

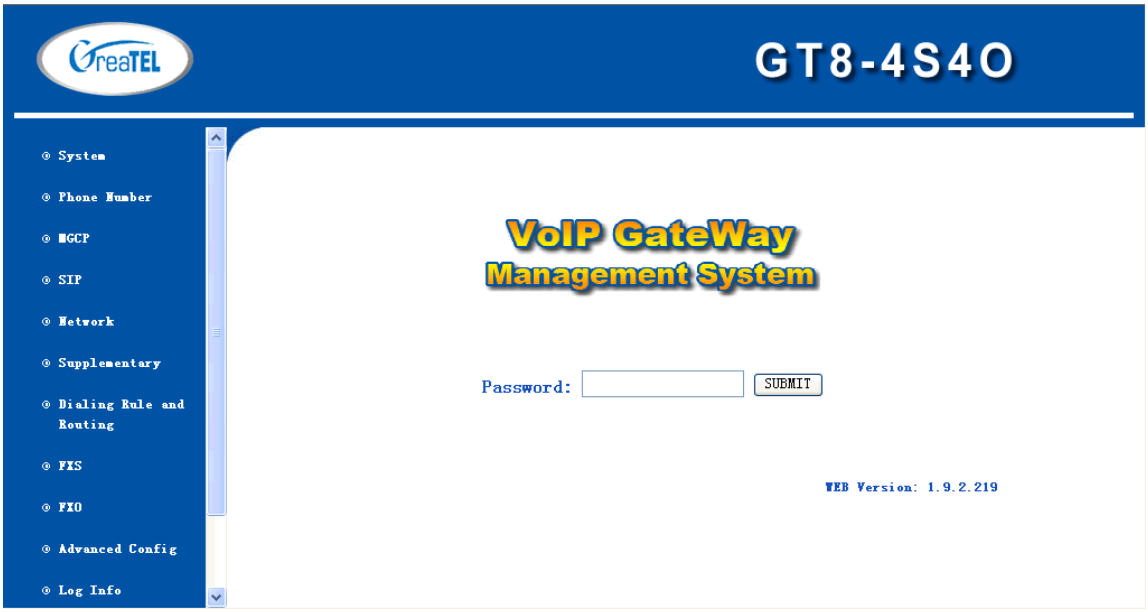


Figure 0-1 VoIP Gateway System Configurations Interface

GT8 has two levels of management：the administrator level (default password：GTadmin) and the operator level (default password：operator). Administrator level has higher access privilege, and is allowed to change password for all users at all levels. Operator level has lower access privilege, and certain options are not available including network configurations, password management and restore factory default settings.

GT8 allows multiple users to log on at the same time. Only the first user logged on with highest privilege is able to change configurations. The rest can only monitor configurations.

Note1：After a user logs on, he/she will be automatically logged off if there is no activities for more than 10 minutes. After that, a user needs to log on again.

Note2：After complete configuration, a user must completely log out instead of just closing the browser. This will elevate the access level of the next logged on user so he/she will be able to change the configurations.

Function Description of Most Used Buttons

At the bottom of each configuration page you will see two buttons: Submit and Default.

- **Submit**: When you are done with configuration, click this button once so that the configuration can be saved. After each submission, you will be prompted by “Submission is successful. Please restart the gateway!” You need to click OK to confirm the action.
- **Default**: Click the button once to restore the factory default setting for each parameter.

Note: Clicking this button only restore the defaults settings for the current page. It is different from System Tools -> Restore Factory Default in that the latter restore the default settings for the whole system.

When the restoration is successful, you will be prompted by “The settings are successfully restored. Please restart the gateway!” You need to click OK to confirm the action.

System Configurations

Click System Configuration link on the left of Figure 4-1, and you will see what is shown in Figure 4-2.

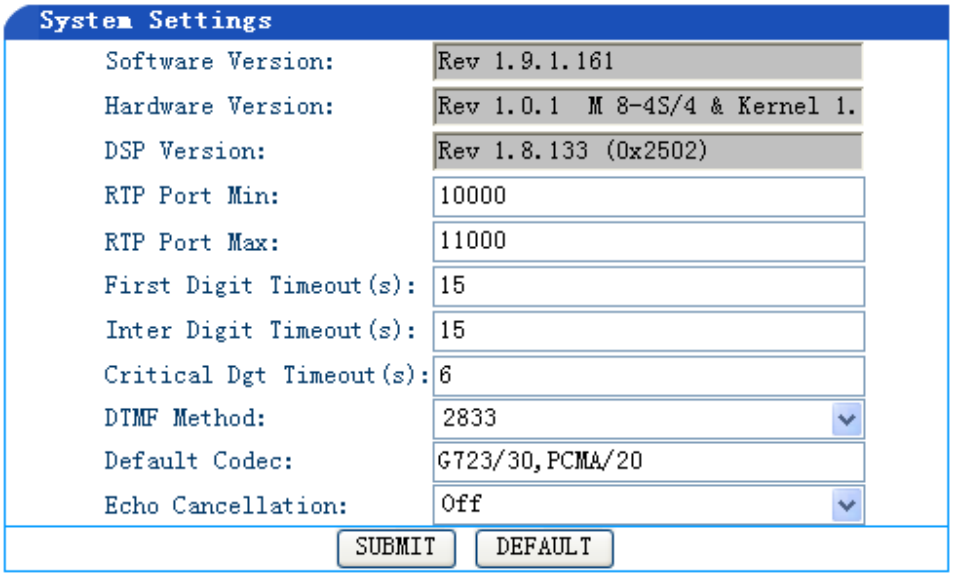


Figure 0-2 System Configuration Interface

Software Version

The **Software Version** field value is automatically detected. You do not need to change this field.

Hardware Version

The **Hardware Version** field value is automatically detected. You do not need to change this field.

DSP Version

The **DSP Version** field value is automatically detected. You do not need to change this field.

RTP Port Min and Max

In the RTP Port Min field enter the minimum value of sending and receiving RTP port. This is a required field. It is recommended that you enter a value that is greater than 10000.

In the RTP Port Max field enter the maximum value of sending and receiving RTP port. This is a required field. It is recommended that you enter a value that equals “2 x number of lines + the minimum value”.

Note: A VoIP call uses two RTP ports: one for RTP and the other for RTCP. If GT8 has four lines (FXS) then the RTP port is set to eight ports at least. If RTP has less than eight ports, four lines can not be used at the same time. GT8 supports up to 8 FXS. So it is highly recommended you set RTP to 16 ports. The default minimum value is 10010～10030. You do not need to change it.

First Digit Timeout

In the **First Digit Timeout** field enter the time (in second) allowed for the dialing of the first digit. When a line goes off-hook, if within the time specified here the first digit has not been dialed, GT8 will treat this as an abandoned call and will indicate to the caller to place the phone on hook. The default value is 12 seconds.

Inter Digit Timeout

In the **Inter Digit Timeout** field enter the time (in second) allowed for the dialing of the middle digits. Counting from the last digit dialed, if within the time specified here no digit has been dialed, the system will send the dialed digits out. The default value is 12 second.

Critical Dgt Timeout

In the **Critical Dgt Timeout** field enter the time (in second) for finished dialing. This parameter is used in conjunction with x.T in the dialing rule. After the first digit in the rule has been dialed, if within the time specified here no digit follows, GT8 will send the dialed number out. The default value is 5 seconds.

DTMF Mode

In the **DTMF [1] Mode** field select the transmission mode. This parameter is used to set DTMF signal transmission mode. Options are Audio mode, 2833 mode, and INFO mode. The default setting is Audio mode.

a) Audio mode is a transparent transmit mode;

b) INFO mode is information transmit mode;

c) 2833 mode is a RTP data packet transmit mode.

Default Codec

In the **Default codec[2]** field select the codec GT8 supports. GT8 support G729A/20, G723/30, PCMU/20, PCMA/20, GSM, iLBC codec

[1] **DTMF（Dual Tone Multi-Frequency）**
In PSTN service, after a call is connected, user's touch tone info is transmitted via DTMF, also known as second dial tone information. It is widely used in intelligent network and value-added services.
• Audio: Voice data transparent transmit mode.
• 2833: A special RTP packet. PT field of the header indicates this is a DTMF packet. See FTC 2833 for details.
• INFO: Optional way of DTMF transmission. As in SIP messages, use INFO to indicate a DTMF signal.

[2] **Voice CODEC**
Also called a "voice codec" or "vocoder," it is a hardware circuit that converts the spoken word into digital code and vice versa. It comprises the A/D and D/A conversion and compression technique. If music is encoded with a speech codec, it will not sound as good when decoded at the other end. A speech codec is an

as well as manifold encoding modes at the same time. Multiple values are demarked by commas. When manifold encoding mode is selected, the gateway will process the communication by selecting the encoding mode front to back, which is supported by both sides.

Table 0-1 Codes supported

Codec supported by GT8	Codec mode	Time interval of RTP packets transmission(unit: ms)
G729A/20	G.729A	20
G723/30	G.723	30
PCMU/20	G.711	20
PCMA/20	G.711	20
iLBC/30	iLBC	30
GSM/20	GSM	20

Echo cancellation

In **the Echo cancellation**[3] select **on** to invoke echo cancellation and **off** to close echo cancellation. The manufacturer's default is on. You do not need to change it.

Set up the Phone Numbers

Click Phone Number link on the left of Figure 4-1, and you will see what is shown in Figure 4-3:

audio codec designed for human voice. By analyzing vocal tract sounds, a recipe for rebuilding the sound at the other end is sent rather than the soundwaves themselves. The speech codec is able to achieve a much higher compression ratio, which results in a smaller amount of digital data for transmission. When telephones were first digitized in the early 1960s, they generated digital streams of 64 Kbps. Since then, speech CODECS have reduced voice to as little as 5 Kbps and less.

[3] **Echo Cancellation**
The term echo cancellation is used in telephony to describe the process of removing echo from a voice communication in order to improve voice quality on a telephone call. In addition to improving quality, this process improves bandwidth savings achieved through silence suppression by preventing echo from traveling across a network.

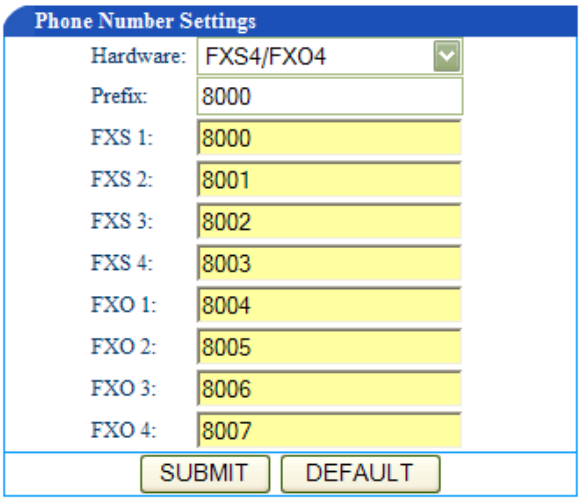


Figure 0-3 Phone Number setting screen

Hardware

Leave the **Hardware** Settings field as it is. GT8 has more than one model, and the model number is set through the software. This parameter is already predefined by the manufacturer. You do not need to change it.

Prefix

In the **Prefix** field enter a prefix number which is for fast setting for serial number. You can leave it blank from FXS2 to FXO4. When FXS1 uses this prefix number, FXS2 uses FXS1 number plus 1, and so on and so forth.

When you set GT8 to MGCP gateway, the value of the prefix should be set to **aaln/0** or **aaln/1**. If MGCP call agent starts from “0”, then use aaln/0; if MGCP call agent starts from”1”, then use aaln/1.

When you set GT8 to SIP gateway, the value of the prefix should be the first number of the serial phone number which the registry server assigns to the gateway. For example if the number of gateway is 2002007, then 200 should be entered in the Prefix field.

FXS(1~4)/ FXO(1~4)

For the **FXS** lines, when the number is not a serial number or a serial number that is not incrementing at order, you can manually enter the number for each FXS line. This gives user more flexibility.

Under MGCP mode

You can set the phone numbers either like shown in Figure 4-4a or in Figure 4-4b：

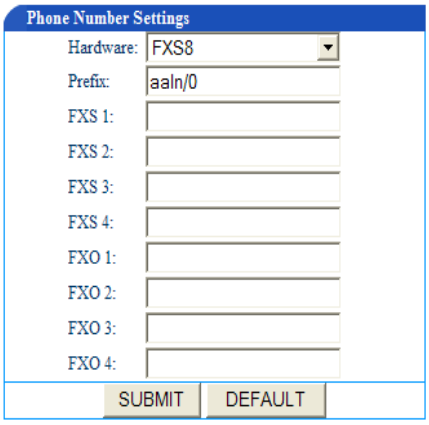


Figure 0-4 a

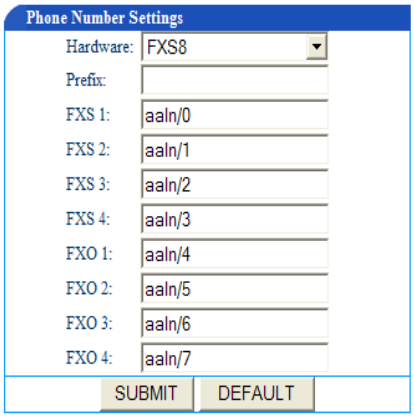


Figure 0-4 b

Under the SIP mode

You can set the phone numbers either like shown in Figure 4-5a or in Figure 4-5b：

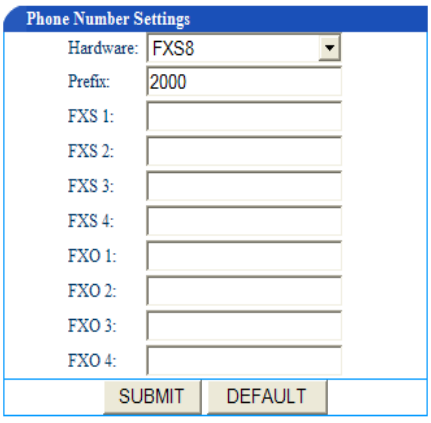


Figure 0-5a

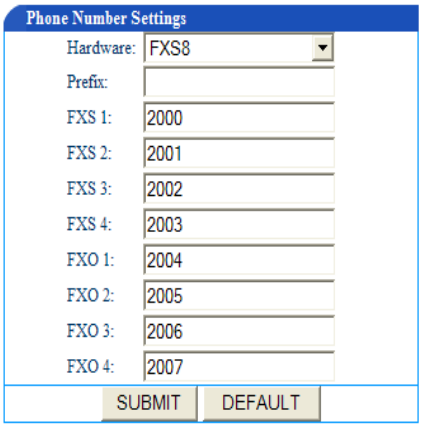


Figure 0-5 b

MGCP Setting

Click the **MGCP Config** link on the left of Figure 4-1. You will see Figure 4-6.

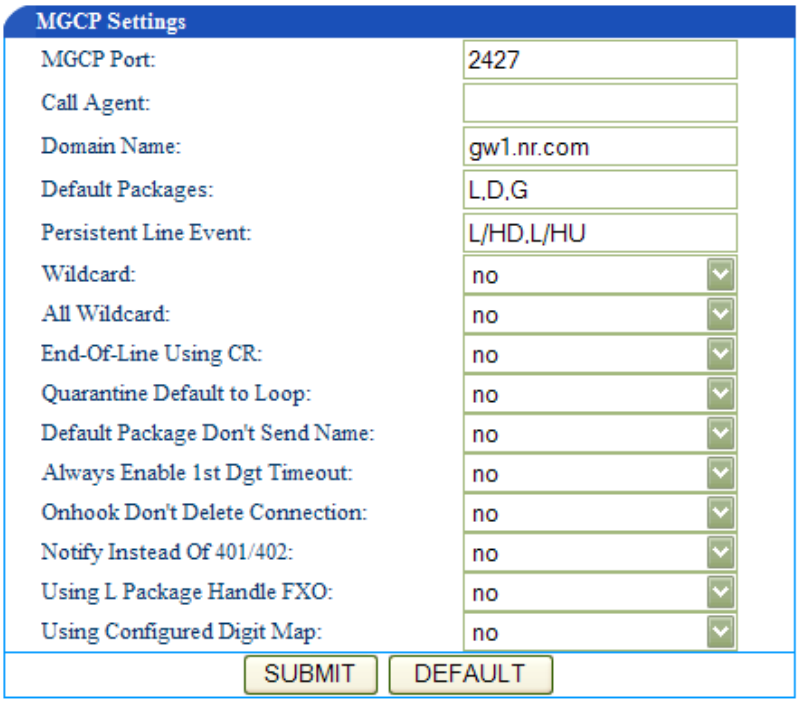


Figure 0-6 MGCP setting screen

MGCP Port

In the **MGCP Port** field enter GT8 gateway MGCP port number (example: 2427). You can use any port number as long as it is not the same as other port numbers.

Call Agent

In the **Call Agent**[4] field enter the call agent address and port number. Address and port number should be separated by **:**. Address could

[4] **Call Agent**
Call Agent, also known as Media Gateway Controller, controls the Media Gateway. In MGCP, a call agent primarily handles all the call processing by linking with the IP network through constant communications with an IP signaling device, for example an SIP Server or an H.323 gatekeeper.
Call Agent is comprised of the call control "intelligence" and a media gateway boasting the media functions, for example conversion from TDM voice to Voice over IP.

Media Gateways feature endpoints for the Call Agent to create and manage media sessions with other multimedia endpoints. Endpoints are sources and/or sinks of data that can be physical or virtual. For creating

be IP address or domain name. If you use domain name, you should invoke DNS service and set parameter of DNS server in the Network Config page. A complete sample configuration is like this: **202.202.2.202:2727**; **callagent.com:2727**.

Domain Name

In the **Domain Name** field enter the internet address or the IP address.

Default Packages

In the **Default Packages** field enter all default packages. Use comma to separate each package. The default setting is L,D,G, which means Line Package, DTMF Package, and Generic Media Package.

Persistent Line Event

In the **Persistent Line Event** field enter all types of persistent line event. Use comma to separate each line event. The gateway will report to call agent when it handles an event. The default setting is L/HD, L/HU, and L/HF. L/HD means off-hook; L/HU means on-hook; and L/HF means hookflash.

Wildcard

In the **Wildcard** field select **yes** or **no** to indicate if GT8 will put the fixed prefix when it registers (such as： aaln/*).

All Wildcard

---

physical endpoints, hardware installation is needed while virtual endpoint can be created using available software.

Call Agents come with the capability of creating new connections, or modify an existing connection. Generally, a media gateway is a network element which provides conversion between the data packets carried over the Internet or other packet networks and the voice signals carried by telephone lines. The Call Agent provides instructions to the endpoints to check for any events and - if there is any - create signals. The endpoints are designed in such a way as to automatically communicate changes in service state to the Call Agent. The Call Agent can audit endpoints and the connections on endpoints.

In the **All Wildcard** field select **yes** or **no** to indicate if GT8 will put the fixed prefix when it registers (such as ： *). If Wildcard and All Wildcard are **yes**, the gateway will deal with all wildcard.

End-Of-Line Using

In the **End-Of-Line Using CR** field select **yes** or **no** to indicate if GT8 will use CR as line stop symbol when sending messages. If set to **no**, CRLF will be used.

Quarantine Default to Loop

In the **Quarantine Default to Loop** field select **yes** or **no** to indicate how GT8 will handle events when there is no response for requirements. If set to **yes**, gateway will report continuously all events of this requirements when it receive a requirement; if set to **no**, gateway will response only once for each requirement.

Default Package Don't Send Name

In the **Default Package Don't Send Name** field select **yes** or **no**. If set to **yes**, the gateway will reply to the default package without a package name; if set to **no** it will reply to the default package with a package name.

Always Enable 1st Digit Timeout

In the **Always Enable 1st Digit Timeout** field select **yes** or **no** to indicate how GT8 will handle events when there is no timeout during the required time. If set to **yes**, the gateway will report timeout according to the settings when the caller does not dial a phone number after going off-hook.

Onhook don't Delete Connection

In the **Onhook don't Delete Connection** field select **yes** or **no**. If you select **yes**, the gateway will delete the connection when the caller does not go on-hook; if you select **no** the gateway will wait for the call agent to delete the connection.

Notify Instead of 401/402

In the **Notify Instead of 401/402**[5] select **yes** or **no**. If you select **yes**, the gateway will use notification message instead of 401/402 message.

Using L Package Handle FXO

In the **Using L Package Handle FXO** field select **yes** or **no**. If you select **yes**, the gateway will treat FXO as FXS; if you select **no**, it will handle FXO and FXS in different ways.

Using Configured Digit Map

In the **Using Configured Digit Map** field select **yes** or **no**. If you select **yes**, the gateway will invoke the dialing rule; if you select **no**, it will use the rule of soft-switch.

SIP Setting

Click the SIP[6] link on the left of Figure 4-1，and the SIP Settings screen displays.

[5] **401/402:** Response Code.

[6] **SIP (Session Initiation Protocol)**
Session Initiation Protocol (SIP) is the Internet Engineering Task Force's (IETF's) standard for multimedia conferencing over IP. SIP is an ASCII-based, application-layer control protocol (defined in RFC 2543) that can be used to establish, maintain, and terminate calls between two or more end points.

Like other VoIP protocols, SIP is designed to address the functions of signaling and session management within a packet telephony network. Signaling allows call information to be carried across network boundaries. Session management provides the ability to control the attributes of an end-to-end call.

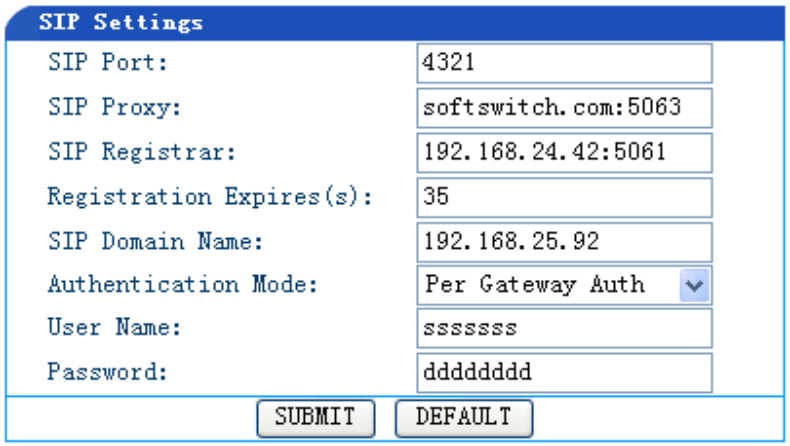


Figure 0-7 SIP settings screen

SIP Port

In the **SIP Port** field enter the number of SIP local port. The default value is 5060. Local port number could be set at will, as long as it doesn't conflict with the other port numbers in the system.

SIP Proxy

In the **SIP Proxy** field enter the address and port number of the Proxy. The address and port number is separated by a colon. The address can be in either IP address form or domain name form. When adopting domain name form, it is necessary to invoke DNS service in the "Network Setting" page and set the parameter of DNS server. The complete and valid setting is as following: **201.30.170.38:5060** and **softswitch.com:5060**.

SIP Registrar

In the **SIP Registrar**[7] field enter the address and port number of the SIP Registrar The address and port number are separated by a colon.

Like other VoIP protocols, SIP is designed to address the functions of signaling and session management within a packet telephony network. Signaling allows call information to be carried across network boundaries. Session management provides the ability to control the attributes of an end-to-end call.

[7] **Registrar**
When a client powers on, it will tell network its IP address in order to be found. We call this procedure "register". The server that accepts this request is called "registrar".

The address can be in either IP address form or domain name form. When adopting domain name form, it is necessary to invoke DNS service in the “Network Setting” page and set the parameter of DNS server. The complete and valid setting is as following: **201.30.170.38:50**60 and **regster.com:5060**.

Registration Expires(s)

In the **Registration Expires(s)** [8] field enter the valid time (in second) for SIP re-registration. The default value is 30 seconds.

SIP Domain Name

In the **SIP Domain Name** field enter the SIP domain name. If the field is left empty, GT8 will use the address of the Proxy as the domain name. It is recommended that you do not use a private network IP address in this field.

Authentication Mode

In the **Authentication Mode** field use the drop down menu to make a selection. **Per Endpoint** means to register and authenticate wholly according to each individual line; **Per Gateway Reg** means to register and authenticate wholly according to gateway; **Per Gateway Auth** means to register according to each individual line, and to authenticate wholly according to gateway.

User Name

Set the **User Name** field if registered as **Per Gateway Reg** or **Per Gateway Auth**; if registered as **Per Endpoint**, do not set this parameter.

Password

[8] **Registration Expires**
In order to control client side, every register message has a certain stored period. If the message is modified in that period, which mean it works for user otherwise Registrar will consider the message is not useful any more, so it will be deleted.

In the **Password** field enter soft-switch authentication password, which can be digits or characters. The password is case sensitive. If registered as **Per Gateway Reg** or **Per Gateway Auth** you need to set this parameter; if registered as **Per Endpoint**, you do not need to set this parameter.

Network Configuration

Click the **Network Config** link on the left side of Figure 4-1. The Network Settings screen displays:

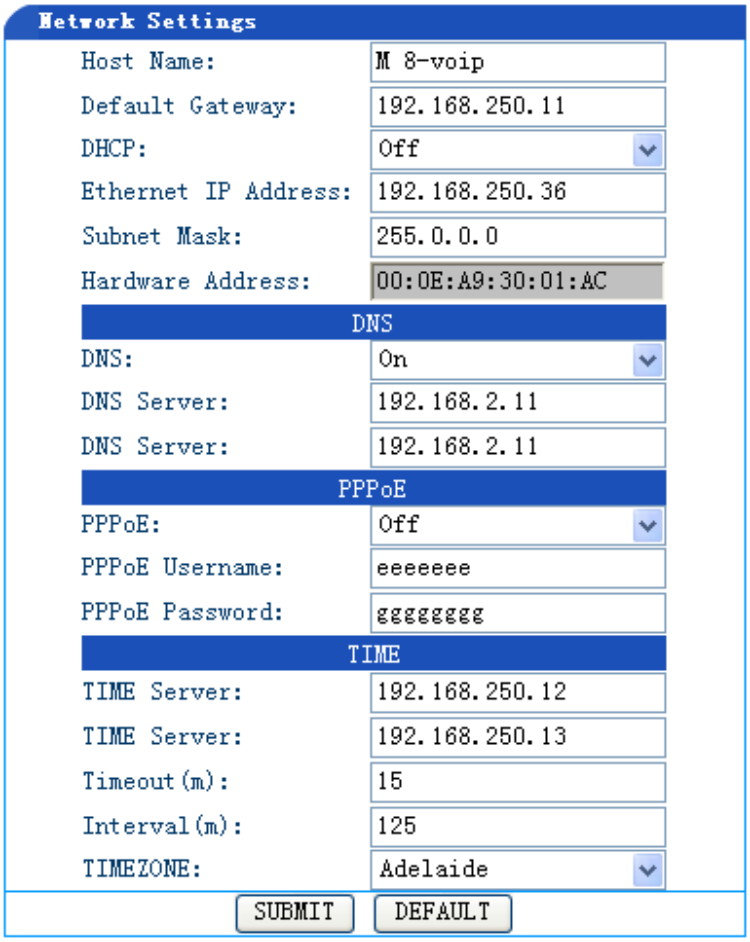


Figure 0-8 Network Settings Screen

Hostname

In the **Hostname** field, enter the GT8 gateway name. You can use your own naming convention according to your network setup.

Gateway IP Address

In the **Gateway IP Address** field enter the default GT8 IP address if you have not enabled DHCP services.

DHCP

In the **DHCP** field select on or off to indicate to use DHCP or not to assign IP addresses and other network settings.

Ethernet IP Address

In the **Host IP Address** field enter the GT8 Ethernet port number if you have not enabled DHCP services. If you have enabled DHCP services, this field will display the IP address that DHCP captures.

Subnet Mask

In the **Subnet Mask** field enter the subnet mask address you obtain from your system administrator or from your ISP if you have not enabled DHCP services.

Hardware Address

Leave the **Hardware Address** as it is. You are not allowed to change it.

DNS

In the **DNS** field select on or off to indicate to turn on DNS services or not. You need to turn on DNS service when you use the domain name as the proxy server address or registration server address in your MGCP or SIP configuration.

DNS Primary Server

In the **DNS Primary Server** field enter GT8 primary DNS server address if you have turned on DNS services.

DNS Alternate Server

In the **DNS Alternate Server** field enter alternate GT8 DNS server address if you have turned on DNS services.

PPPoE

In the **PPPoE** field select **on** or **off** to indicate to use PPPoE service or not.

- If you selected **on** in Step 11, you need to enter your user name in the **PPPoE Username** field.
- If you selected **on** in Step 11, you need to enter your password in the **PPPoE Password** field.

Time Primary Server

In **Time Primary Server** field enter the IP address of your primary Time server.

Time Alternate Server

In **Time Alternate Server** field enter the IP address of your alternate Time server.

Timeout

In the **Timeout** field enter the time (in minute) allowed to locate the Time server. If the server is not located within the time allowed, GT8 will try to locate it again.

Interval

In the **Interval** field, enter the time interval (in minute) at which GT8 will synchronize its time with the Time server.

Time Zone

In **Time Zone** field select the GT8 location.

The following is the options are available for this parameter: Midway, Honolulu, Anchorage, Tijuana, Denver, Mexico_City, Indianapolis, Glace_Bay, Buenos_Aires, South_Georgia, Cape_Verde, London, Amsterdam, Cairo, Moscow, Muscat, Karachi, Almaty, Bangkok, Beijing, Tokyo, Canberra, Magadan, Auckland, Newfoundland, Tehran, Kabul, Calcutta, Adelaide

Supplementary Features

The features in this section are enabled only when using SIP Protocol. The same features are provided by the proxy server when using MGCP Protocol. There is no need for configuration.

Setting up the Feature Keys

Click the **Supplementary** link on the left side of Figure 4-1. Then click **Feature Code**. The Feature Code Settings screen displays (see Figure 4-9). You can set up all the supplementary feature keys from here. The general rule is *xx for enable (i.e. dial the * key plus any two digits that represent the feature) and #xx for disable (i.e. dial the # key plus any two digits that represent the feature). The screen shows all the features with their default values. You can replace the default with any numbers you like.

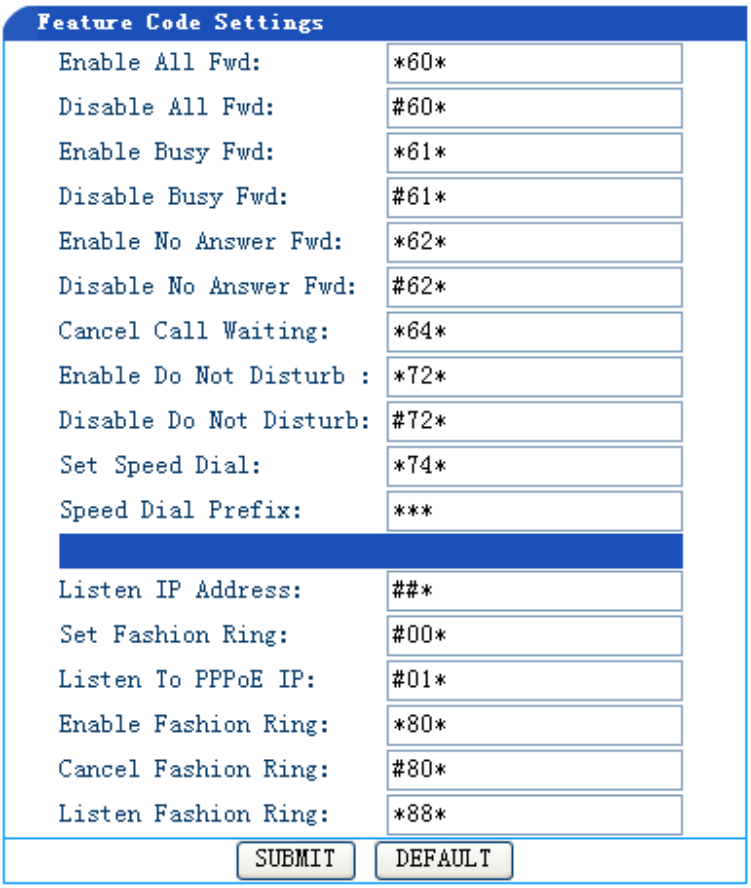


Figure 0-9 Feature Code Setting Screen

Enable All Forwarding

This allows the customer to define and enable forwarding all calls function. The default function key is *60. To use this feature the customer must first sign up for the call forwarding service.

Disable All Forwarding

This feature allows the customer to disable the All Forwarding service.

For example:

To forward all calls to phone number 5614888888, the enabling key is *60. The disenabling key is #60.

a) To enable:

Go off hook → Dial *60 → Upon hearing the dialing tone, enter 5618888888 → Dial # to end → Go on hook.

b) To verify:

Go off hook → Dial *60 → Upon hearing the forwarded number dial # to end → Go on hook.

c) To disable:

Go off hook → Dial #60 → Go on hook.

Enable Busy Forwarding

This allows the customer to enable the forwarding feature when the line is busy. The default function key is *61. To use this feature the customer must first sign up for the call forwarding service.

Disable Busy Forwarding

This allows the customer to disable Busy Forwarding function. The default function key is #61.

For example:

To forward all the calls when the line is busy to phone number 5614601688, the enabling key is *61. The disenabling key is #61.

a) To enable:

Go off hook → Dial *61 → Upon hearing the dialing tone, enter 5614601888 → Dial # to end → Go on hook.

b) To verify:

Go off hook → Dial *61 → Upon hearing the forwarded number dial # to end → Go on hook.

c) To disable:

Go off hook → Dial #61 → Go on hook.

Enable No Answer Forwarding

This allows the customer to define and enable the forwarding feature when the line is busy. To use this feature the customer must first sign up for the call forwarding service.

Disable No Answer Forwarding

The default function key for this feature is #62.

For example:

To forward calls to 5618881680 when nobody is answering the calls, the enable key is *62, and the disable key is #62.

a) To enable,

Go off hook → Dial *62 → On hearing the dialing tone, dial 5618881680 → Dial # to end → Go on hook.

b) To verify,

Go off hook → Dial *62 → On hearing the forwarded number go on hook.

c) To disable,

Go off hook → Dial #62 → Go on hook.

Cancel Call Waiting

This allows the customer to disable the call waiting function when a call is in progress to avoid interruption. The default function key is *64. This feature works for only one call. To completely remove call waiting, please refer to section 4.9.

Enable Do Not Disturb

When this feature is enabled, the customer will not hear the ringing tone when a call comes in. The caller will hear busy tones. The default function key is *72. To use this feature, the customer needs to first sign up for the Do Not Disturb services. Please refer to section 4.9.

Disable Do Not Disturb

This will restore the normal call handling. The default function key is #72.

Set Speed Dial

The default function key is *74. This allows the customer to use a two-digit code (from 20 to 49) for dialing the complete digits. To use this feature the customer needs to sign up for speed dial services.

Speed Dial Prefix

This defined the identifiers for speed dial. The default function key is **. Before using the speed dial, the customer must first dial these two digits.

For example:

The speed dial code for phone number 5613221680 is 20, and the speed dial prefix is ** .

a) To enable speed dial,

Go off hook → Dial *74 → On hearing the dialing tone, dial 20 plus 5613221680 → Dial # to end.

b) To verify,

Go off hook → Dial *74 → On hearing the dialing tone, dial 20 plus * to end → On hearing the complete digits, go on hook.

c) To use the speed dialing,

Go off hook → Dial ** plus 20.

d) To disenable,

Go off hook → Dial *74 → On hearing the dialing tone, dial 20 plus # to end.

Listen IP Address

This allows the customer to hear the IP address of his phone line. The default function key is ##.

Enable Line Search

This allows the customer to hear the phone number of this his phone line. The default function key is #00.

Listen to PPPoE IP Address

This allows the customer to hear the gateway PPPoE IP address. The default function key is #01.

Enable Fashion Ring

This allows the customer to set the ring tones to his liking. The default function key is *80.

Cancel Fashion Ring

This restores the ringing tone to normal. The default function key is #80.

For example:

Use the default function key for enabling distinctive ring. Set the distinctive ring ID number from 01 (must have two digits).

a) To enable,

Go off hook → Dial *80 → On hearing the dialing tone, dial 01 → Go on hook.

b) To verify,

Go off hook → Dial *80 → On hearing the distinctive ringing go on hook.

c) To disenable,

Go off hook → Dial #80 → Go on hook.

Listen Fashion Ring

The default function key is *88.

To use:

Go off hook → Dial *88

→ Dial distinctive ring ID number 01 → Listen to the ring tones

→ Dial distinctive ring ID number 05 → Listen to the ring tones

→ Dial distinctive ring ID number 12 → Listen to the ring tones

→ Dial distinctive ring ID number 34 → Listen to the ring tones

→ ... ...

→ Go on hook.

Set up All Forward

Click the **Supplementary** link on the left side of Figure 4-1. Then select **Set Forward All**. You will see All Forward Settings screen, as shown in Figure 4-10:

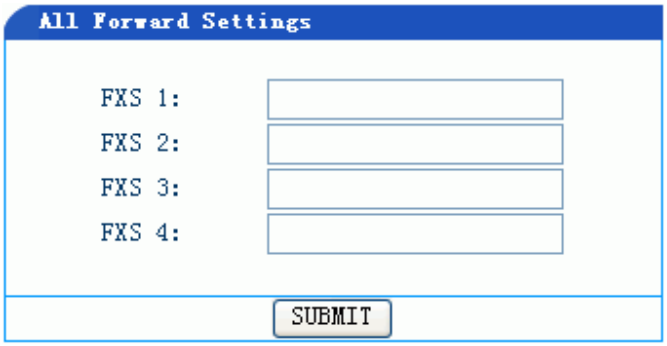


Figure 0-10 All Forward Settings Screen

This screen is used to enter the forwarding numbers for All Forward feature subscribers or to display during operation the forwarding numbers entered by the end users. Calls will be forwarded to those numbers only when the end users sign up for the Call Forwarding services and when they enable this feature using the function key.

Set up Busy Forward

Click the **Supplementary** link on the left side of Figure 4-1. Then click **Busy Forward**. The Busy Forward Settings screen displays, as shown in Figure 4-11:

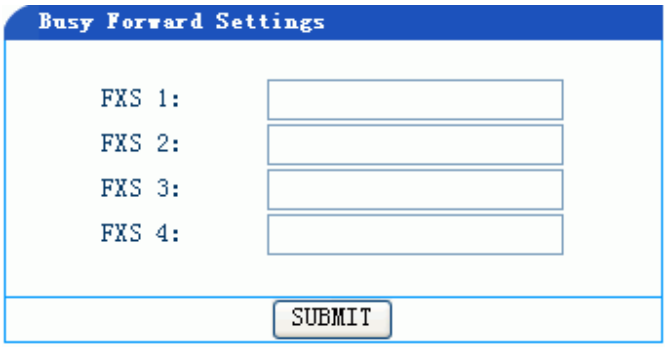


Figure 0-11 Busy Forward Settings Screen

This screen is used to enter forwarding numbers for Busy Forward feature subscribers or to display during operation the forwarding numbers entered by the end users. Calls that come on busy lines will be forwarded to those numbers only when the end users sign up for the Call Forwarding services and when they enable this feature using the function key.

Set up No Answer Forward

Click the **Supplementary** link on the left side of Figure 4-1. Then click **No Answer Forward**. The No Answer Forward Settings screen displays, as shown in Figure 4-12:

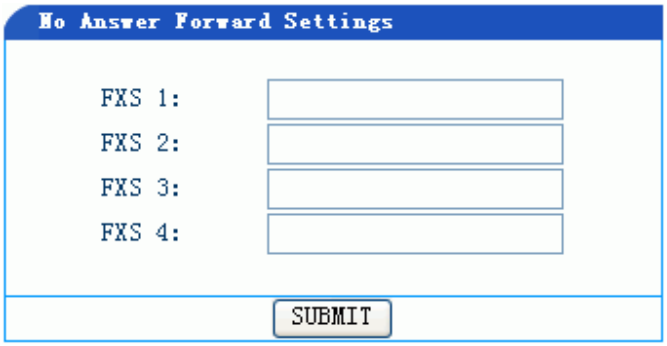


Figure 0-12 No Answer Forward Setting Screen

This screen is used to enter forwarding numbers for No Answer Forward feature subscribers or to display during operation the forwarding numbers entered by the end users. Calls that get no answers will be forwarded to those numbers only when the end users

sign up for the Call Forwarding services and when they enable this feature using the function key.

Set up Fashion Ring

Click the **Supplementary** link on the left side of Figure 4-1. Then click **Fashion Ring**. The Fashion Ring Settings screen displays, as shown in Figure 4-13:

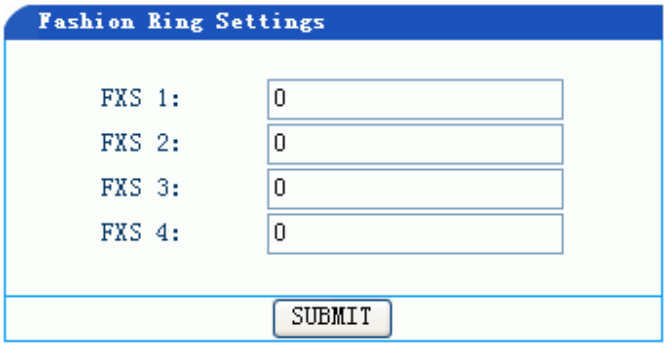


Figure 0-13 Fashion Ring Settings Screen

This screen is used to enter distinctive ring serial numbers for Fashion Ring feature subscribers or to display during operation the fashion ring numbers entered by the end users. End users that have singed up for Fashion Ring services and have enabled the feature using the function key will hear the distinctive rings.

Set up Hotline

Click the **Supplementary** link on the left side of Figure 4-1. Then click **Hotline**. The Hotline Settings screen displays, as shown in Figure 4-14:

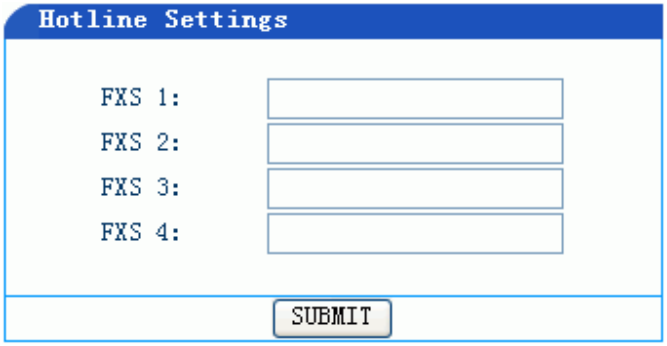


Figure 0-14 Hotline Settings Screen

This screen is used to enter hotline numbers for Hotline feature subscribers. End users that have singed up for Hotline or Delay Hotline services have access to this feature.

For example, FXS Line 1's hotline number is set up as1680. If the end user of Line 1 has signed up for Hotline services, when he goes off hook the phone will automatically dial 1680. However hotline users cannot dial any other numbers. If the end user of Line 1 also signed up for Delay Hotline services, within six seconds of going off hook, if no other number is dialed, the hotline number 1680 will be dialed; if another number is dialed within six seconds, then this call is treated as a normal call. Hotline function will be ignored.

Dialing Plan and Routing Table

Set up the Dialing Plan

Click the **Dialing Plan** link on the left side of Figure 4-1. Then click **Digit Map**. The Digit Map Rules screen displays, as shown in Figure 4-15:

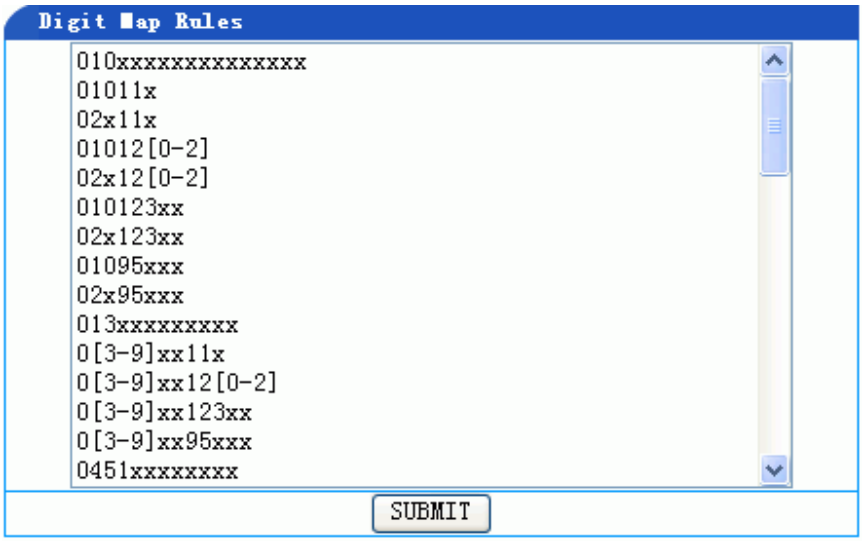


Figure 0-15 Digit Map Rules Screen

Digit Map is used to determine if the digits received are the complete numbers dialed, so that the dialing process will terminate and the digits will be sent out in a speedy way. This can shorten the connection time for calls.

GT8 has in its default Dialing Plan most of the domestic digit map rules. You do not have to re-configure them. You can add new rules when necessary. The following is an illustration of the common rules:

Table 0-2 Common Digit Map Rules

X	Any single digit between numbers 0 to 9.
.	any multiple digit between numbers 0 to 9.
##	terminate dialing after receiving two digits. ## is GT8 gateway's default function key for listening to the IP address.
x.T	The gateway will check a number of any lengths that is composed of any number between 0 and 9. If no new digits are received within the "dialing finish" time, the gateway will send out the detected number.
x.#	a number of any length that starts with any number between 0 and 9. If the end user dials # right after the number, GT8 will stop number reception and send out the number before #.
***xx**	terminate dialing after receiving * plus any two digits. *xx is mainly used to enable the supplementary services (such as Distinctive Ring, Do Not Disturb, and Call Forwarding).
#xx	end dialing after receiving # plus any two digits. #xx is mainly used to disable the supplementary services.
[2-8]xxxxxx	a seven-digit number that starts with any number between 2 and 8. This is used to terminate local call dialing.
02xxxxxxxx	an 11-digit number that starts with 02. This is used to terminate long distance call dialing that starts with 02.
013xxxxxxxx	a 12-digit number that starts with 013. This is used to terminate long distance cellular calls that start with 013.
13xxxxxxxx	an 11-digit number that starts with 13. This is used to terminate local cellular calls that start with 13.
11x	a three-digit number that starts with 11. This is used to terminate emergency calls.
9xxxx	a five-digit number that starts with 9. This is used to terminate special service calls
17911	send out the number right after receiving 17911. This serves as an example of terminating a special number.

Set up the Routing Table

Click the **Dialing Plan** link on the left side of Figure 4-1. Then click **Route Table**. The Route Table screen displays, as shown in Figure 4-16:

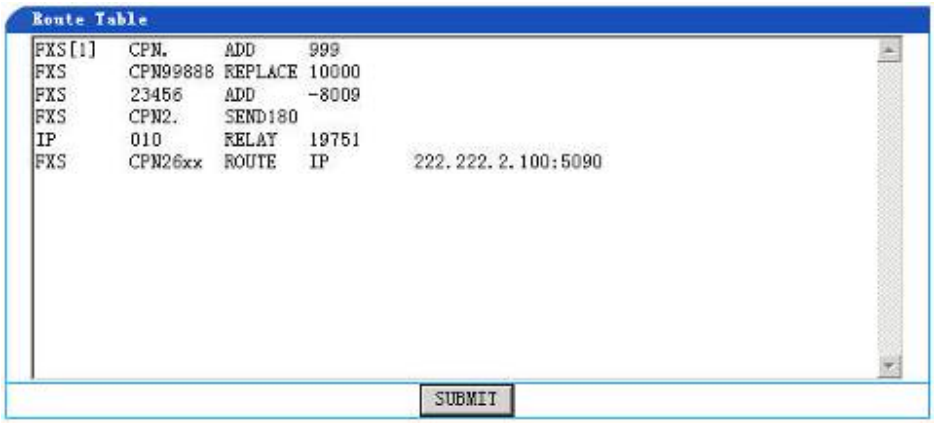


Figure 0-16 Routing Table Screen

Routing table serves two main functions: number swapping and route exchange. The table is executed from top to bottom. Number swapping always has advantage over route exchange. A routing table can have a maximum of 100 entries.

Note: The routing table is empty by default. All the calls go to the SIP Proxy server, and are routed by this server.

1. Number Swapping

One phone number consists of three sections: Origination, Number, and Action.

- **Origination** can have the following values: IP, FXS, and FXO. IP can be any IP address, a specified IP address, and a specified IP address plus the port number. FXS and FXO can be a specific line number (for example FXS1, FXO2 or FXS 1 – 2, etc.)
- **Number** can be the calling number, or the called number. Default is the called number. If it is the calling number, add CPN before the number as the identifier. The number can use any digit between 1 to 9, *, ., #, X etc, just like the digit map. The common rules are:
  - Numbers, such as 114, 68640585
  - The beginning digits of a number, such as 61xxxx, or 612x, or 61
  - Expressions such as 268[0-1, 3-9], which indicates a number that starts with 268 and followed by any number from 0 to 1 or 3 to 9
  - The search for a matching number follows the principle of “shortest and quickest”. For example, x

equals all numbers; xx equals all two-digit numbers; 12x equals all three-digit numbers that start with 12

- **Action** defines the processing method and the actual information that has been processed. It can have three values:
  - **KEEP**: Keep means to keep the number. Another number goes after it. If that number is positive, it means to count the number from the front; if the number is negative, it means to count the number from the backward. For example,
    FXS 02168640585 KEEP -8.

    This means to keep the last eight digits of this called number from the FXS, that is 68640585.
  - **REMOVE**: Remove means to remove the number. Another number goes after it. If that number is positive, it means to count the number from the front; if the number is negative, it means to count the number from the backward. For example,

    FXS 021 REMOVE 3.

    This means to remove 021 if the called number from an FXS starts with 021
  - **ADD**: Add means to add digits before or after the called number. Another number goes after it. If that number is positive, it means to add before the number; if the number is negative, it means to add after the number. For example,

    FXS1 CPNX ADD 021
    FXS2 CPNX ADD 010

    This means to add 021 to all the CNP from FXS1; to add 010 to all the CPN from FXS2.

    Another example:
    FXS CPN6120 ADD -8888,

    meaning to add 8888 to CPN from the FXS that start with 6120
  - **REPLACE:** means to replace the number, followed by the number to be replaced to. For example, FXS CPN88 REPLACE 2682000, meaning for a CPN

from an FXS that starts with 88, replace it with 2682000

- **REPLACE:** cross replacement, means to replace the number corresponding to the called number or calling number

  For example,

  FXS 12345 REPLACE CPN-1/8621
  This means remove the last one digit then add 8621 at the end of the calling number corresponding to the called number from an FXS that starts with 12345

  FXS CPN13 REPLACE CDPN0/0
  This means add 0 at the head of the called number corresponding to the calling number from an FXS that starts with 13.

  FXS 12345 REPLACE CPN1/
  This means remove one digit at the head of the calling number corresponding to the called number from an FXS that starts with 12345

  FXS 12345 REPLACE CPN/5678
  This means add 5678 at the head of the calling number corresponding to the called number from an FXS that starts with 12345

  FXS 12345 REPLACE CPN-/5678
  This means add 5678 at the end of the calling number corresponding to the called number from an FXS that starts with 12345

  Notes:

  - CPN and CDPN: the appointed digit stream.
  - If CPN and CDPN behind REPLACE, the CPN means calling number and the CDPN means called number.
  - If the number before REPLACE, CPN means calling number, the called number is indicated with digits directly.
  - If no “/”, means replace all.
  - “-“: means the end of the number.

- CPNn: n means how many digits will be replaced.
- If n is 0, means insert
- If n larger than the quantity of whole digits, means replace all.
- If there is no number at the end of "/", means remove
- If there is no number at the front of "/", means insert, the gateway will insert 0 automatically.

- **END:** means to terminate certain number processing. When performing number swapping from top to bottom, if END or ROUTE is present, then end number swapping. For example,

  FXS 12345 ADD -8001
  FXS 12345 REMOVE 4
  FXS 12345 END

  This means for the called number from an FXS that starts with 12345, first add 8001 at the end of the number; then remove the first four digits; and end the number swapping for CDN that starts with 12345.Another example,

  IP[222.34.55.1] CPNX. REPLACE 2680000
  IP[222.34.55.1] CPNX. ROUTE FXS 2

  This means for any CPN of any lengths that comes from IP address 222.34.55.1, replace it with 2680000, and then route it to the second line of the FXS.

- **SEND180**, meaning to force sending 180. For example,
  FXS CPN2 SEND180,

  meaning for CPN from the FXS that starts with 2, send 180.

- **SEND183**, meaning to force sending 183. For example,
  FXS CPN3 SEND183,

  meaning for CPN from the FXS that starts with 3, send 183.

- **CODEC** means the encoding and decoding method of the CPN/CDN, followed by the actual name of the codec. For example

  PCMU/20/16.

  PCMU is the codec method; 20 meaning every 20ms; 16 is the length of echo cancellation. If echo cancellation is not enabled, a 0 will append at the end automatically (like PCMU/20/0), indicating echo cancellation is disabled. For example,

  IP 6120 CODEC PCMU/20/16

  This means for CDN from an IP address and starting with 6120, use codec PCMU/20. Echo cancellation length is 16ms.

- **RELAY** is one function of IP dialing. For example,

  IP 010 RELAY 17909

  This means for CDN from an IP address and starting with 010, dial 17909 first.

2. Route Exchange

One route consists of five sections: Origination, Number, Action, Destination, and Destination Information. Routing table routes the number from an origination to the destination.

- **Origination** can have the following values: IP, FXS, and FXO. IP can be any IP address, a specified IP address, and a specified IP address plus the port number. FXS and FXO can be a specific line number (for example FXS1, FXO2 or FXS 1 – 2, etc.)
- **Number** can be the calling number, or the called number. Default is the called number. If it is the calling number, add CPN before the number as the identifier. The number can use any digit between 1 to 9, *, ., #, X etc, just like the digit map. The common rules are:
  - Numbers, such as 114, 68640585
  - The beginning digits of a number, such as 61xxxx, or 612x, or 61
  - Expressions such as 268[0-1, 3-9], which indicates a number that starts with 268 and followed by any number from 0 to 1 or 3 to 9

  - The search for a matching number follows the principle of “shortest and quickest”. For example, x equals all numbers; xx equals all two-digit numbers; 12x equals all three-digit numbers that start with 12

- **Action** should be ROUTE, meaning to route a call.
- **Destination** can have the following values: NONE, IP, FXS, and FXO.
  - Routes that have IP as the Origination usually have FXO, FXS, or NONE as Destination
  - Routes that have FXO or FXS as the Origination usually have IP or NONE as Destination
  - Routes that have FXX/FXS as Destination can use the Destination Information as the route or to hunt for an idle line
  - Routes that have IP as Destination: the Destination Information section must provide a specific gateway IP address plus its SIP port number (if no port number is defined, use the default port number 5060). For example: *192.168.2.10:5066*

    If the IP address is local, use format localhost:5060 or 127.0.0.1:5060. For example,

    IP 8621 ROUTE FXS 1
    IP CPN8620 ROUTE FXS 2

    This means a call to the called number from an IP address that starts with 8621 will be routed to the first FXS line; while a call with calling number that starts with 8620 will be routed to the second FXS line.

    Another example:

    FXS 021 ROUTE IP 228.167.22.34:5060
    FXS 020 ROUTE IP 61.234.67.89:5060

    This means a call to the called number from an FXS that starts with 021 will be routed to IP address 228.167.22.34; while a call from an FXS that starts with 020 will be routed to IP address 61.234.67.89.

    IP CPN[1, 3-5] ROUTE NONE

This means a call from an IP address with calling number that start with 1, 3, 4, and 5 will not be routed.

Set up the FXS Ports

This is applicable only to GT8 that have FXS ports.

One GT8 can have up to eight FXS lines. Each line is configured the same way. You can customize the configuration according to real life situation. The following is a sample configuration.

Click the **FXS Config** link on the left side of Figure 4-1. Then click **FXS 1**. The FXS Settings screen displays, as shown in Figure 4-17:

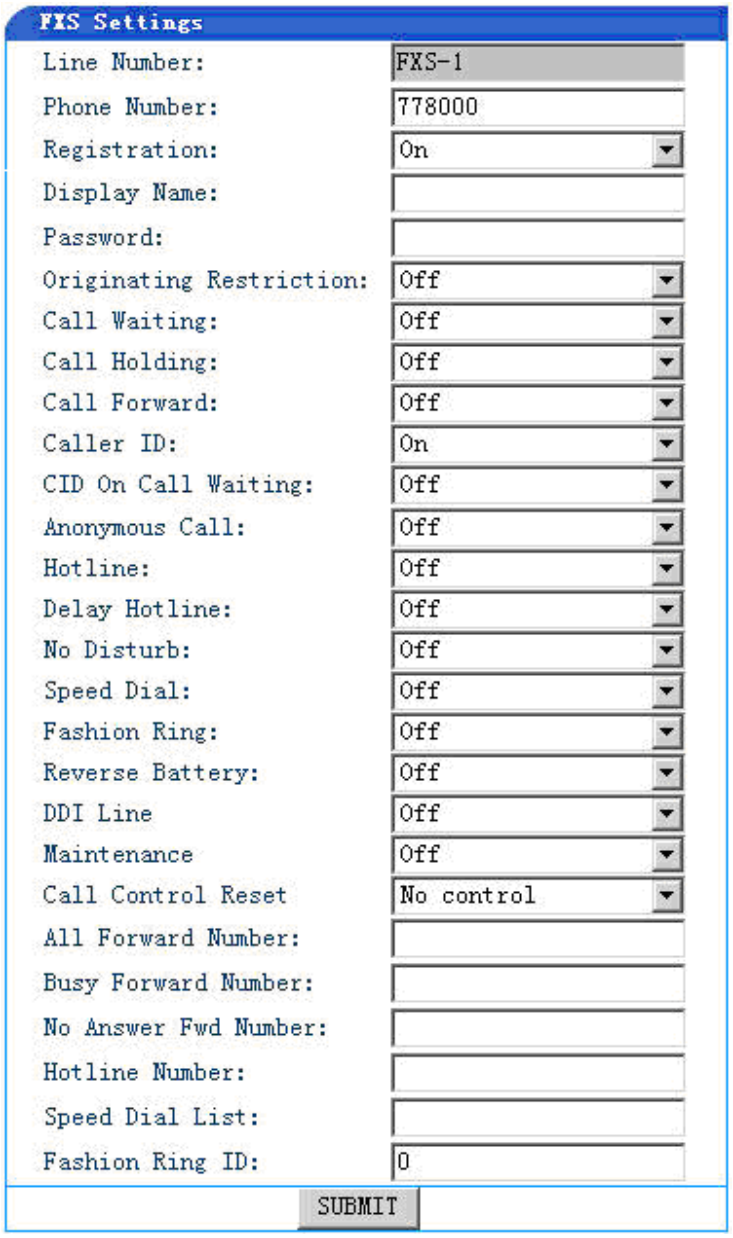


Figure 0-17 FXS Settings Screen

Phone Number

In **Phone Number** field enter the phone number that is set up in section 4.3.

Registration

In **Registration** field, select **on** (to register) or **off** (not to register).

Display Name

In **Display Name** field enter the content to display in the outgoing calls. You can enter up the 30 characters. FXS lines that have name display capability can display what is entered here.

Password

In **Password** field enter the registration password if you selected **on** in Step 3.

Note：The functions beyond this point only apply to SIP protocol. When using MGCP protocol, there is no need to set them up, as the set up does not work.

Originating Restriction

Select **on** (to indicate the line can only receive calls but not initiate calls) or **off** (no restriction).

Call Waiting field

Select **on** (enable) or **off** (disable).

Call Holding

Select **on** (enable) or **off** (disable).

Call Forward

Select **on** (enable) or **off** (disable).

Caller ID

Select **on** (enable) or **off** (disable).

CID On Call Waiting

Select **on** (enable) or **off** (disable).

Anonymous Call

Select **on** (enable) or **off** (disable).

Hotline

Select **on** (enable) or **off** (disable).

Hotline Delay

Select **on** (enable) or **off** (disable).

Do Not Disturb

Select **on** (enable) or **off** (disable).

Speed Dial

Select **on** (enable) or **off** (disable).

Fashion Ring

Select **on** (enable) or **off** (disable).

Reverse Battery[9]

Select **on** (enable) or **off** (disable). If on, the line will send a reverse signal upon call connection and the accounting system starts fee calculation.

[9] **Reverse Battery Signaling**
Loop signaling in which battery and ground are reversed on the tip and ring of the loop to give an "off-hook "signal when the call receiver answers. Note: Reverse-battery signaling may be used either for a short period, or for the duration of a call, to indicate that it is a toll call.

DDI Line

Select **on** (enable) or **off** (disable) to use the Direct Dialing In (DDI) function. Default is **off**.

Maintenance

Select **on** or **off** to indicate to turn the power on/off for this line. Default is **off**.

Call Control Reset

Caller Control (Calling Party Control) is to indicate that the "Calling Party" has hung up. It will release a line at once. If Called Party has hung up, but Calling Party has not hung up, it will release FXS line between 60 seconds and 180 seconds according to the set up in **Release Timeout**

No Control

All Forward Number

In **All Forward** field, the number set up in section Set up All Forward will display. You can also enter a different number here to overwrite the previous number.

Busy Forward

In **Busy Forward** field, the number set up in section Set up Busy Forward will display. You can also enter a different number here to overwrite the previous number.

No answer Fwd Number

In **No Answer Fwd Number** field, the number set up in section 错误！未找到引用源。will display. You can also enter a different number here to overwrite the previous number.

Hotline Number

In **Hotline Number** field, the number set up in section Set up Hotline will display. You can also enter a different number here to overwrite the previous number.

Speed Dial List

In **Speed Dial List** field, enter the Speed Dial Code number (any two digits between 20 and 49) plus the actual number. Multiple entries are separated by */*. Example: 20-3221860/21-7558888/22-5552525.

Fashion Ring ID

In **Fashion Ring ID** field, the number set up in section Set up Fashion Ring will display. You can also enter a different number here to overwrite the previous number.

Set up the FXO

This is applicable only to GT8 that has FXS ports.

One GT8 can have up to four FXO lines. Each line is configured the same way. You can customize the configuration according to real life situation. The following is a sample configuration.

Click the **FXO Config** link on the left side of Figure 4-1. Then click **FXO 1**. The FXO Settings screen displays, as shown in Figure 4-18:

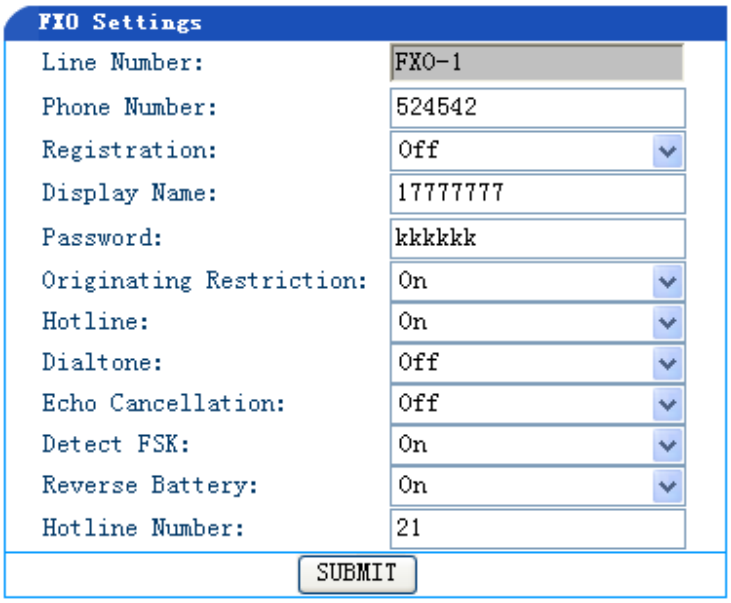


Figure 0-18 FXO Setting Screen

Phone Number

In **Phone Number** field enter the phone number that is set up in section 4.3.

Registration

In **Registration** field, select **on** (to register) or **off** (not to register).

Display Name

In **Display Name** field enter the content to display in the outgoing calls. You can enter up the 30 characters. FXO lines that have name display capability can display what is entered here.

Password

In **Password** field enter the registration password if you selected **on** in Step 3.

Note： The functions beyond this point only apply to SIP protocol. When using MGCP protocol, there is no need to set them up, as the set up does not work.

Originating Restriction

In **Originating Restriction** field, select **on** (to indicate the line can only receive calls but not initiate calls) or **off** (no restriction).

Hotline

In **Hotline** field, select **on** (enable) or **off** (disable).

Dialtone

In **Dialtone** field, select **on** (enable) or **off** (disable). This function is disabled once the Hotline function is on.

Echo Cancellation

In **Echo Cancellation** field, select **on** (enable) or **off** (disable).

Detect FSK

In **Detect FSK** field, select **on** (enable) or **off** (disable). This indicates to check and forward the calling number from PSTN or not.

Hotline Number

In **Hotline Number** field, the Hotline number set up in Set up Hotline will display. You can also enter a different number here to overwrite the previous number.

Advanced Options

System Advanced Options

Click the **Advance Config** link on the left side of Figure 4-1. Then click **System Config**. The System Optional screen displays, as shown in Figure 4-19:

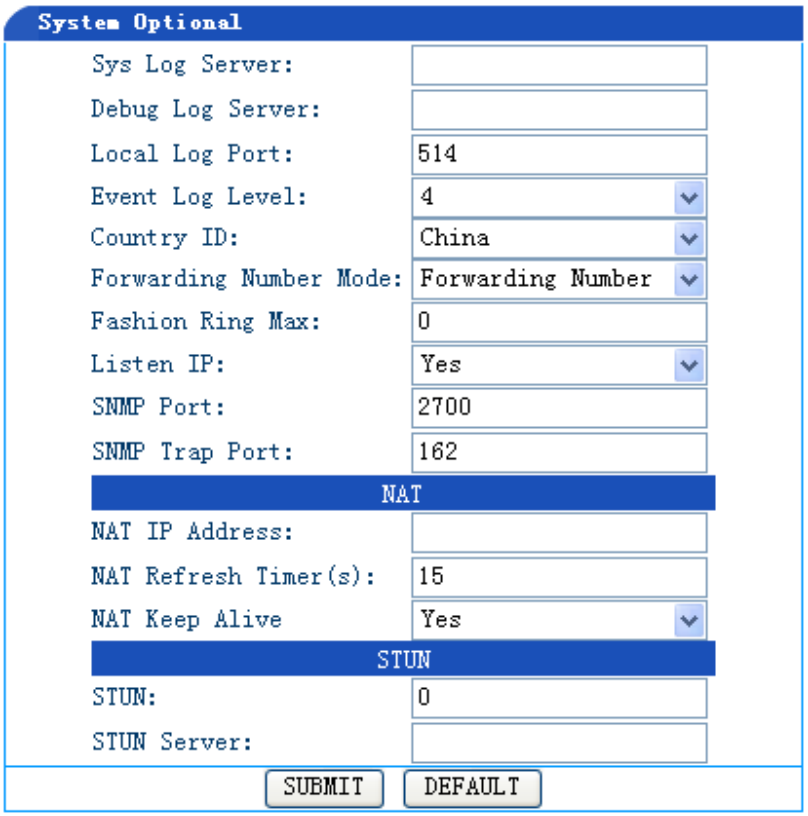


Figure 0-19 System Optional Screen

Sys Log Server

This is the IP address of the Event Log Server. It is used for remote debugging. You do not need to set it under normal circumstance.

Debug Log Server

This is the IP address of the Debug Log Server. It is used for remote debugging. You do not need to set it under normal circumstance.

Local Log Port/ Event Log Port

Default is 514.

Event Log Level

Select any number from 1 to 5. The higher the level, the more detailed the log. Default is set to **3**. Higher level may slow down system performance.

Country ID

In **Country ID** field, select the country in which the gateway is operated，and gateway will adopt different disposals according to different countries' standards.

Forwarding Number Mode

In **Forwarding Number Mode** field, use the pull down menu to select Calling Party Number or Forwarding Number. This determines if the calling party number or the forwarding number should be displayed in the last line. For example, if line 3221680 has call forwarding function and the forwarded number is 7558888, when line 5552525 calls 3221680, line 7558888 will display 5552525 if Calling Party Number is selected here; if Forwarding Number is selected, then line 7558888 will display 3221680.

Fashion Ring Max

In **Fashion Ring Max** field, enter the maximum Fashion Ring file size and the highest Fashion Ring ID number.

Listen IP

Select yes (to allow the customer to hear the IP address by push ##) or no (not to allow).

SNMP Port

In **SNMP Port** field, enter the UDP port used by Simple Network Management Protocol. SNMP provides a way to collect network management information from network equipments as well as a way for the equipments to report problems and errors to the network.

SNMP Trap Port

In **SNMP Trap Port** field, enter the UDP port used by SNMP Trap command. The default value is 162. TRAP is one command of SNMP, whose main function is to send alarm asynchronously to network management workstation, notifying it that some event that fulfills the proposition has occurred.

Nat IP Address

If gateway is within the private network and the network outside the NAT[10] is public, you can map the IP obtained from SDP[11] message to a fixed IP. No default value

Note: You can search the IP address from the following websites: www.ipchicken.com; www.showmyip.com; www.whatismyip.com; www.myipaddress.com; and wwww.whatismyipaddress.com.

Nat Refresh Time

In **Nat Refresh Time** field, enter the time interval in seconds to refresh NAT status. This request is send to the STUN Server. This value is used when NAT Alive is enabled or when requesting STUN services.

Nat Keep Alive

Select **yes** (to keep it alive) or **no** (not to keep it alive).

STUN

Select **on** to turn on STUN[12] service or **off** to turn off STUN service.

STUN Server

In **STUN Server** field, enter the IP address of the STUN Server. A STUN server can send requests as well as generate responses. STUN server normally runs in public network and therefore is stateless. If this field is empty, the default STUN server will be used.

---

[10] **NAT (Network Address Translator)**
Network Address Translation, an Internet standard that enables a local-area network (LAN) to use one set of IP addresses for internal traffic and a second set of addresses for external traffic. A NAT box located where the LAN meets the Internet makes all necessary IP address translations.

[11] **SDP (Session Description Protocol)**
SDP describes multimedia sessions for the purpose of session announcement, session invitation and other forms of multimedia session initiation.

[12] **STUN (Simple Traversal of UDPover NATs)**
STUN（Simple Traversal of UDP over NATs）
A protocol that allows applications to detect that a network address translation (NAT) is being used. It can also detect the type of NAT and IP address assigned by it. STUN was developed to support interactive, two-way communications over the Internet such as for voice (VoIP) and videoconferencing. The STUN client sends requests to a STUN server, which is typically hosted by the service provider.
Unlike application layer gateways (ALGs) and Middlebox Communications (MIDCOM), which also support two-way communications through NATs, STUN requires no changes to the NAT.

Advanced FXO Options

Click the **Advance Config** link on the left side of Figure 4-1. Then click **FXO Config**. The FXO Optional screen displays, as shown in Figure 4-20:

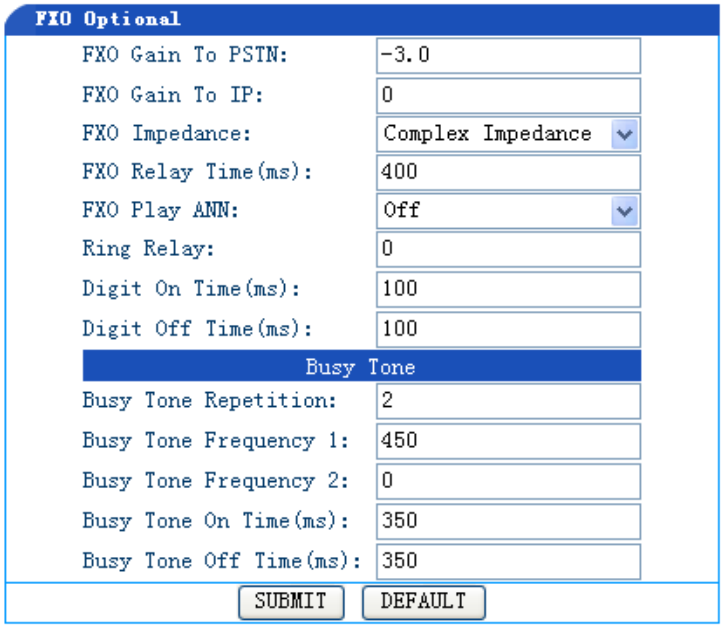


Figure 0-20 FXO Optional Screen

FXO Gain[13] To PSTN

Enter the volume increase into the PSTN. Value range is -6 - +3dB. Default is -3.5dB.

FXO Gain To IP

Enter the volume increase into the IP. Value range is -3 - +3dB. Default is 0.

FXO Impedance[14]

[13] **Sending Gain (or signaling gain)**
When detected signals are not strong enough over the network, we use signal gain parameter to increase the strength of the signal.

[14] **Line Impedance**
A measure of the total opposition to current flow in an alternating current circuit, made up of two components, ohmic resistance and reactance, and usually represented in complex notation as Z = R + iX, where R is the ohmic resistance and X is the reactance. It also refers to an analogous measure of resistance to an alternating effect, as the resistance to vibration of the medium in sound transmission.

Select an FXO impedance number. Options include Complex Impedance; 600Ω; and 900Ω. Default is 600Ω.

FXO Relay Time

In the **FXO Relay Time** field, enter the delay time in sending out the digits to the PSTN from the FXO side. The default is 400ms.

FXO Play ANN

FXO hotline to FXS, when FXS is busy, FXO can play announcement.

On: play, **Off** : not play. Default value is off.

Ring Relay

Select to enable (**1**) or disable (**0**) the function of relay the ring to FXS

Digit On Time

Enter any number from 80 to 150. This parameter specifies the signaling mode[15] of auto dialing from FXO to PSTN. The default is 100ms.

Digit Off Time

In the **Digit Off Time** field, enter any number from 80 to 150. This is the interval at which FXO sends out digits. The default is 100ms.

Busy Tone Repetition

In **Busy Tone Repetition** field, enter a number from 2 to 5. This is the number of times GT8 keeps checking busy tone before it takes further action.

Busy Tone Frequency 1

In **Busy Tone Parameter 1** field, enter one of the signal frequency parameters (IS is 61485). The formula is: frequency=[65536*cos(2*PI*f/8000)] where the value of **frequency** is an integer and **f** is the actually frequency value.

Busy Tone Frequency 2

[15] **Signaling Mode**
Signal mode refers to inserting a silence signal periodically into the voice stream. Standard used in China specifies “450 Hz, 350ms ON + 350ms OFF”, means signal cycle is 700ms; insert 350 ms signal plus a silence signal of 350 ms for each cycle.

In **Busy Tone Parameter 2** field, enter another signal frequency parameter (IS is 0). The formula is: frequency=[65536*cos(2*PI*f/8000)] where the value of **frequency** is an integer and **f** is the actually frequency value.

Busy Tone On Time

In **Busy Tone On Time** field, enter the time one busy tone will last. This time should be determined by the equipment the FXO is connected to. International standard is 350ms.

Busy Tone Off Time

In **Busy Tone Off Time** field, enter the time interval each busy tone is sent. This time should be determined by the equipment the FXO is connected to. International standard is 350ms.

Advanced FXS Options

Click the **Advance Config** link on the left side of Figure 4-1. Then click **FXS Config**. The FXS Optional screen displays, as shown in Figure 4-21:

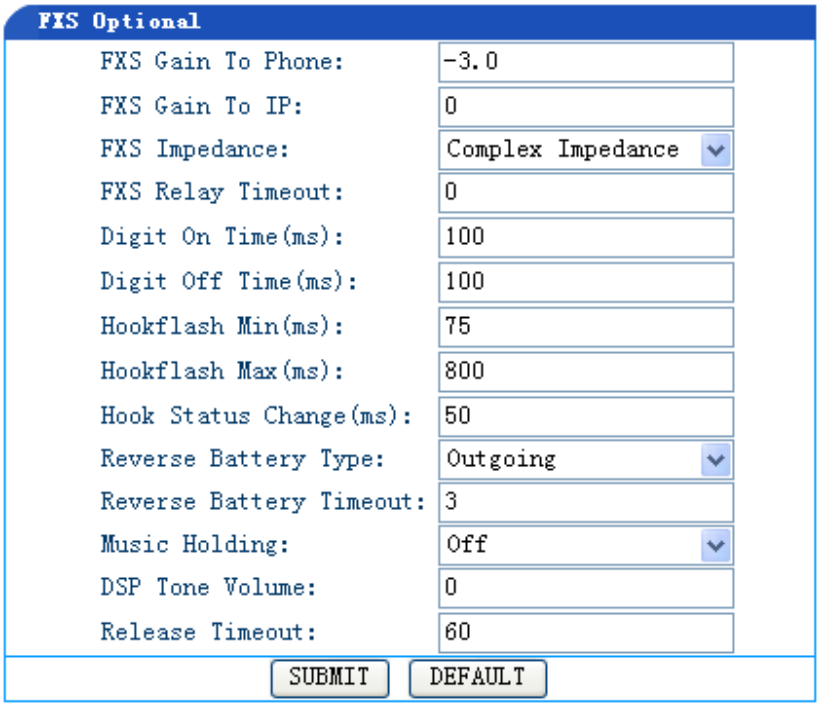


Figure 0-21 FXS Optional Screen

FXS Gain To Phone

Enter the volume increase to FXS telephone. The range is -6 to +3db. Default is -3.0dB (decrease 3 decibel)

FXS Gain To IP

Enter the volume increase to IP network. The range is -3 to +3db. Default is 0dB.

FXS Impedance

Set the FXS impedance number. Options include Complex Impedance; 600Ω; and 900Ω. Default is 600Ω.

FXS Relay Time

In the **FXO Relay Time** field, enter the delay time in sending out the digits to the PSTN from the FXO side. The default is 400ms.

Digit On Time

In the **Digit On Time** field, enter any number from 80 to 150. This is the speed at which FXO will keep at sending out digits. The default is 100ms.

Digit Off Time

In the **Digit Off Time** field, enter any number from 80 to 150. This is the interval at which FXO sends out digits. The default is 100ms.

Hookflash Min

In the **Hookflash Min** field, enter the minimum time for an effective hookflash. Normally this number should be bigger than 75ms.

Hookflash Max

In the **Hookflash Max** field, enter the maximum time for an effective hookflash. Normally this number should be smaller than 800ms.

Hook Status Change

In the **Hook Status Change** field, enter a number from 20 to 1000. This is the hook status change time. If less than the time set here, GT8 will ignore this status change.

Reverse Battery Type

Default value is **Outgoing**. Two options are available for this parameter:

- **Outgoing:** starting the collect call billing after outgoing call is connected;
- **Both:** starting the collect call billing after incoming or outgoing call is connected).

Reverse Battery Timeout

Set the delay from the ringing to sending the collect call billing signal. The default is 3 seconds. The valid value is from 0 to 30 seconds.

Music Holding

Select to enable (**On**) or disable (**Off**) fashion ring when a call is put on hold. Default value is: Off.

Port Volume Control

In the **Volume Control** field, enter the volume gain to the PSTN. The default is -3.5dB.

Release Timeout

FXS line release time when set Caller control mode. The default value is 60 seconds, and the valid range is from 60 to 180.

Advanced IP Options

Click the **Advance Config** link on the left side of Figure 4-1. Then click **IP Config**. The IP Optional screen displays:

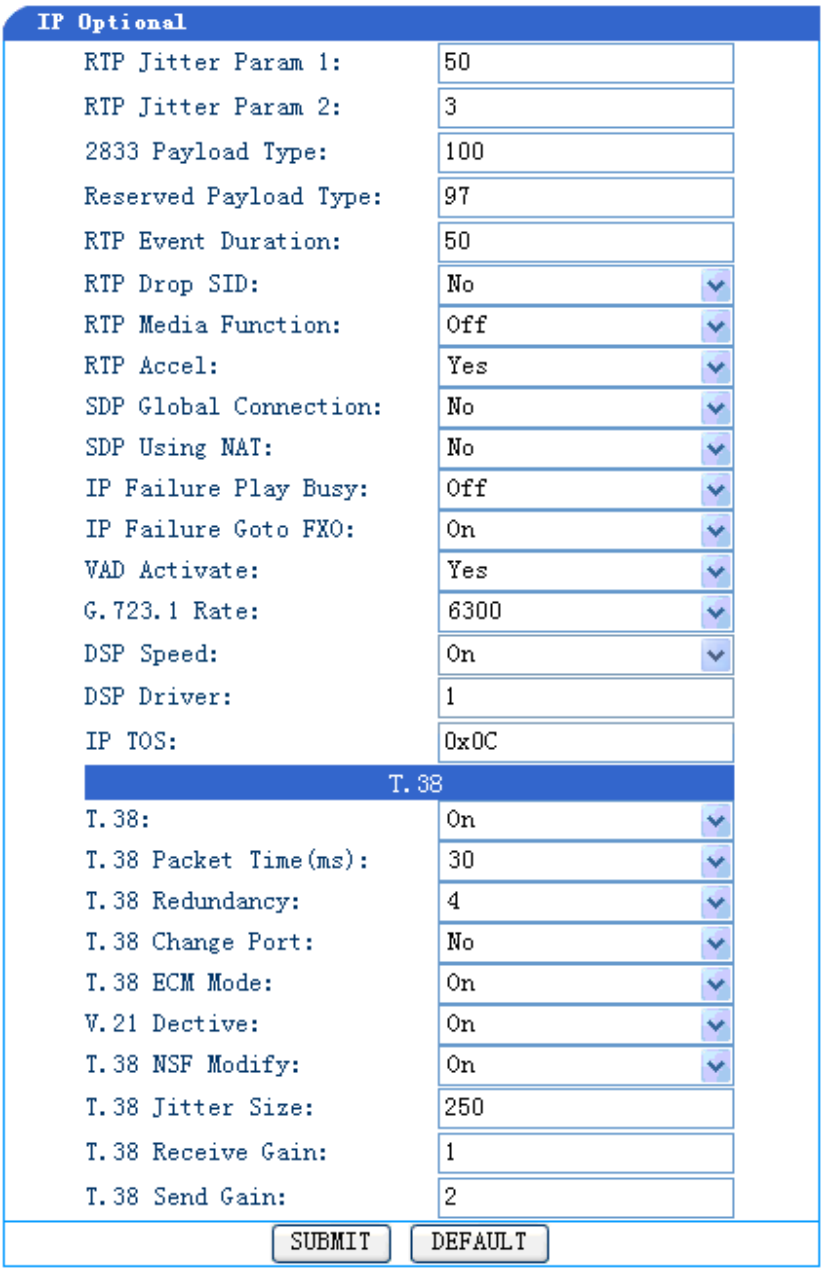


Figure 0-22 IP Optional screen

RTP Jitter Param1

Default is 50. It is recommended that you do not change this value.

RTP Jitter Param2

Default is 3. It is recommended that you do not change this value.

2833 Packet Type

In the **2833 Packet Type** field enter a value from 97 to 127. This parameter is used for transmitting 2833 packet type. The default load type is 97.

Reserved Payload Type

Enter a value from 97 to 127. This is the RTP load type when using the iLBC codec. Default is 97.

RTP[16] Event Duration (ms)

When gateway detects DTMF events, and if RFC2833 is enabled in System Settings, it will send out the RPT event at a regular interval according to the time interval set here. Default value is: 50 ms.

RTP Drop SID[17]

Set if the gateway should ignore received RTP SID. Default value is: No. Note: This needs to be set only when irregular frames are received. RTP SID frames in Irregular lengths may cause noise or weird sound.

- **Yes:** Ignore silence packet.
- **No:** Keep silence packet.

RTP Media Function[18]

Set whether to enable Voice Proxy. Default value is: No. This is more applicable to the setting where one gateway is on the public network while the other is on the private one. Under normal situations, Voice Proxy is not needed. When symmetric RTP function is enabled, gateway checks received RTP packet and extract IP and port information from it before dynamically changing IP address and port number used for sending.

- **On:** Enable Voice Proxy.
- **Off:** Disable Voice Proxy.

RTP Accel

[16] **RTP (Real-time Transport Protocol):** See Glossary.

[17] **SID:** Stands for Silence Information Description.

[18] **Voice Proxy:** See Glossary.

Set whether to apply RTP gain when sending and receiving. Default value is: Yes.

- **Yes:** Enable.
- **No:** Disable.

SDP[19] Global Connection

Setup whether to obtain far end IP address from SDP global connection. Default value is No.

- **Yes:** Get far end IP address from SDP Global Connection Information.
- **No:** Get far end IP address from Connection Information after SDP Media Description.

SDP Using NAT

Setup whether to use NAT address in out-going SDP. Default value is No.

- **Yes:** Use NAT Address in out-going SDP.
- **No:** Use Local host IP address in out-going SDP.

Notes: This parameter works only when gateway is able to get an NAT address. There are two ways to obtain NAT address:

a) When gateway is using STUN function

b) When gateway starts to register and the 200 OK it gets from the registration server contains NAT information.

IP Failure Play Busy

Select **on** or **off** to indicate to send or not busy tones when IP network doesn't connect.

IP Failure Goto FXO

Select **on** or **off** to indicate to switch or not all calls to FXO port when IP network doesn't connect. It is recommended that you set this parameter to on. If IP network doesn't connect, in order to protect router, all IP calls go to PSSTN through FXO.

VAD Active

[19] **SDP(Session Description Protocol)**
SDP describes multimedia sessions for the purpose of session announcement, session invitation and other forms of multimedia session initiation.

In the **VAD Active** field if you select **yes**, the speech packet will not be sent out in the mute of the call, and noise is added to the speech stream to replace the mute. It is recommended that you set this parameter to **yes** to save the network bandwidth.

G.723.1[20] Rate

In the **G.723.1 Rate (BPS)** field enter either 5300 or 6300 according to specific applications.

DSP Speed

On: DSP clock frequency is 225MHz

Off : DSP clock frequency is 200MHz

DSP Driver

Default value is **1** or **0** to indicate to turn on or off dsp driver.

IP Packet in Tos[21]

IP Packet in Tos Field is used to set the quality assurance for the different classes of service.

T.38

In the **T.38** field select **on** or **off** to indicate whether to invoke T.38 fax function or not. **on** means to invoke; **off** means not to invoke.

T.38 Data Frame Length

In the **T.38 Data Frame Length** field set the packing interval for each T.38 data frame. The value can be set as the fallowing: 10/20/30/40/50/60.

T.38 Redundancy Frame Numbers

[20] **G.723.1 Speech Codec**
G.723.1 dual-rate speech coder performs compression and decompression of 8 kHz speech signals. It encodes 16-bit PCM samples into 16-bit code-words yielding 10 or 12 code-words per 240 sample frames for the 5.3 Kbps and 6.3 Kbps channels respectively. 60% of a phone call consists of silence. Silence Compression Scheme and Voice Activity Detection (VAD) reduce network bandwidth usage and save valuable speech resources.

[21] **TOS (Type of Service)**
TOS has 8 bits reserved to the service type in the IP datagram. 0-2 means precedence. 6-7 are unused. 3-5 means D (requests low delay), T (requests high throughput), R (requests high reliability), respectively.

In the **T.38 Redundancy Frame Numbers** field set the number of the T.38 data frame in each T.38 data packet (the effective range is 1 to 6).

T.38 Change to UDP Port

In the **T.38 Change to UDP Port** field select **yes** or **no** to indicate if the gateway will change the UDP port when switching to T.38 mode. If set to **no** it will use the RTP port established during the connection.

T.38 ECM Mode

Select On or Off to indicate whether to invoke T.38 error detection mode. Default value is: Off.

- **On:** Enable error detection mode. When error occurs and is detected, gateway automatically re-sends fax.
- **Off:** Disable error detection mode.

V.21[22] Detective

Setup whether to enable V.21 fax error detection. Default value is: On.

Note: It is not necessary to enable V.21 if fax machine can send normal signals. Set this parameter to Off to reduce DSP processing load.

- **On:** Enable
- **Off:** Disabled

T.38 NSF[23] Modify

Setup whether to shield from non-standard fax transmission. Default value is: On. Recommendation: Set it to on.

- **On:** Shield from non-standard transmission.
- **Off:** Not shield from non-standard transmission.

T.38 Jitter Size[24]

[22] **V.21**
V.21 is an ITU-T recommendation for full-duplex communication between two analogue dial-up modems using audio frequency-shift keying modulation at 300 bauds to carry digital data at 300 bit/s. If fax machine doesn't get any fax signal, the gateway can detect any fax signal throughV.21.

[23] **NSF（Non-Standard facilities）**
Non-standard fax facilities are those whose operation features are not defined by ITU. Some of those features are encoded in FIF but their encoding method was not defined.

Setup T.38 jitter buffer value. Default value is 250ms. Valid value range is 40 ~ 1000.

T.38 Receive Gain[25]

This sets the T.38 receiving gain value. Default value is 1. Valid range is 0 ~ 4.

- **Values 0 and 1:** mean -6dB and -3dB enhancement, respectively.
- **Value 2:** means 0dB gain.
- **Values 3 and 4:** means 3dB and 6dB enhancements, respectively.

T.38 Send Gain

This field sets the T.38 sending gain value. Default value is 2. Valid range is 0 ~ 4.

- **Values 0 and 1:** mean -6dB and -3dB increment, respectively;
- **Value 2:** means 0dB increment.
- **Values 3:** and 4 mean 3dB and 6dB increment, respectively.

Advanced SIP Options

Click the **Advance Config** link on the left side of Figure 4-1. Then click SIP Config. The SIP Optional screen displays:

[24] **Jitter Buffer**
Jitter is a major factor affecting the quality of IP calls. Jitter buffer is a software process that eliminates jitter caused by transmission delays in Internet telephony (VoIP) network. As the jitter buffer receives voice packets, it adds small amounts of delay to the packets so that all of the packets appear to have been received without delays. Voice signals are sequential by nature (i.e., they must be played back in the order in which they were sent) and the jitter buffer ensures that the received packets are in the correct order. Without a jitter buffer to smooth the transmission, data can be lost, resulting in choppy audio signals. There are two types of jitter buffers - dynamic and static. A static jitter buffer is hardware-based and configured by the manufacturer. A software-based jitter buffer is called a dynamic jitter buffer and can be configured by the system or network administrator.

[25] **Signal Gain**
When detected signals are not strong enough over the network, we use signal gain parameter to increase the strength of the signal.

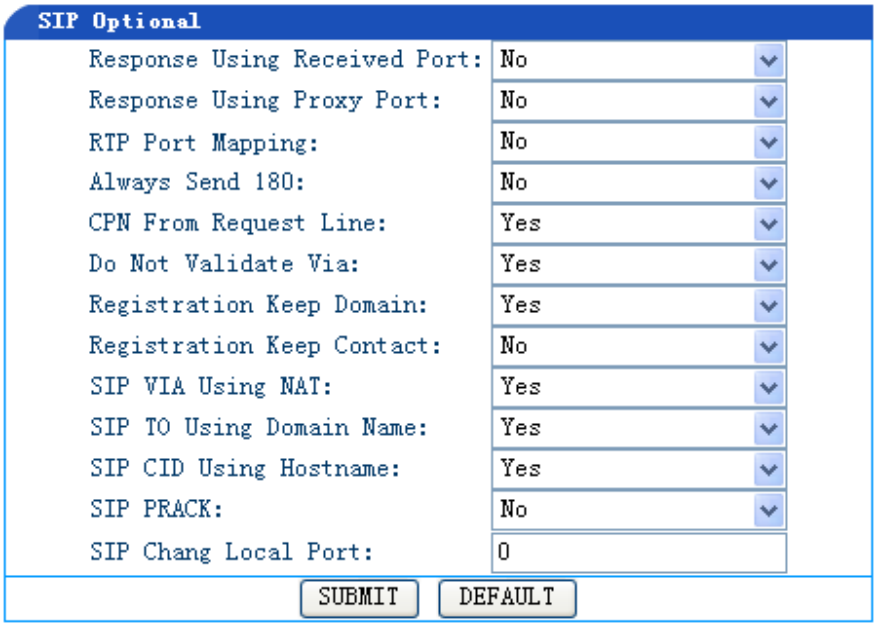

SIP Optional
Response Using Received Port: No
Response Using Proxy Port: No
RTP Port Mapping: No
Always Send 180: No
CPN From Request Line: Yes
Do Not Validate Via: Yes
Registration Keep Domain: Yes
Registration Keep Contact: No
SIP VIA Using NAT: Yes
SIP TO Using Domain Name: Yes
SIP CID Using Hostname: Yes
SIP PRACK: No
SIP Chang Local Port: 0
SUBMIT
DEFAULT

Figure 0-23 SIP Optional screen

Response Default Port Using Received Port

In the **Response Default Port Using Received Port** field use **yes** or **no** to indicate to use received port as the response port or not. **yes** means to use the received port as the response port; **no** means to use the default port 5060.

Response Default Port Using Proxy Port

In the **Response Default Port Using Proxy Port** field use **yes** or **no** to indicate to use proxy port as the response port or not. **yes** means to use the proxy port as the response port; **no** means to use the default port 5060.

RTP Port Mapping

In the **RTP Port Mapping** field use **yes** or **no** to indicate to use invoke RTP port or not. yes means to invoke RTP port mapping function, and use local SIP port and RTP port; no means not to use RTP port mapping function, and use the port requested by STUN.

Always Send 180

In the **Always Send 180** field select **yes** or **no**. **yes** means to send180 in ISDN mode with voice prompt so that user can hear the normal ring back tone instead of voice prompt tone; no means to send18x.

CPN From Request Line

In the **CPN From Request Line** field select **yes** or **no**. **yes** means to get the called number from Request Line; no means to get the called number from To field.

Response Do Not Check Via

In the **Response Do Not Check Via** field use **yes** or **no** to indicate whether to ignore the **Via** field or not. **yes** means to ignore the Via field of the received message; **no** means not to ignore the Via field of the received message.

Registration Keep Domain Name

The **Registration Domain Name Information during** field is applicable only to gateways which use character type domain name. **yes** means to use the full domain name information to register; **no** means to just use the shared part of the domain name to register.

Registration Keep Contact

The **Registration Keep Contact** field is used specially for the registering mode of the gateway during the penetration of the private network. If it is set to yes, the system will keep the original Contact information during registering； otherwise it will return NAT information.

SIP VIA using NAT Information

The **SIP VIA using NAT Information** is used to indicate whether to use the public network address information obtained by NAT or the private network address information when setting up SIP VIA field. If set to **yes**, the gateway will use the public network address information obtained by NAT; otherwise it use the private network address information.

SIP TO Adopt Domain Name Information

The **SIP TO Adopt Domain Name Information** is used to indicate whether to use the Proxy information or the domain name information in SIP Setting when setting up SIP TO field. Yes means the gateway will use domain name information in SIP Setting; No means to use the Proxy information.

SIP CID Using Hostname

Set whether Call ID in SIP message to use server host name or to use IP address. Default value is No.

- **Yes:** Use host name.
- **No:** Use an IP address.

SIP PRACK

The **SIP PRACK** is used to require the receiver to return PRACK thus to provide a reliability mechanism for provisional responses. Set **Yes** (enable) or **No** (disable) with default **No**.

SIP Change Local Port

Select 1 or 0 to turn on or off the function of change local sip port. The default value is 0.

Backup Agent Config

Click **Advanced Config > Backup Agent Config** from the left pane. The following displays:

Note: For information on how to use Submit and Default, see most used buttons.

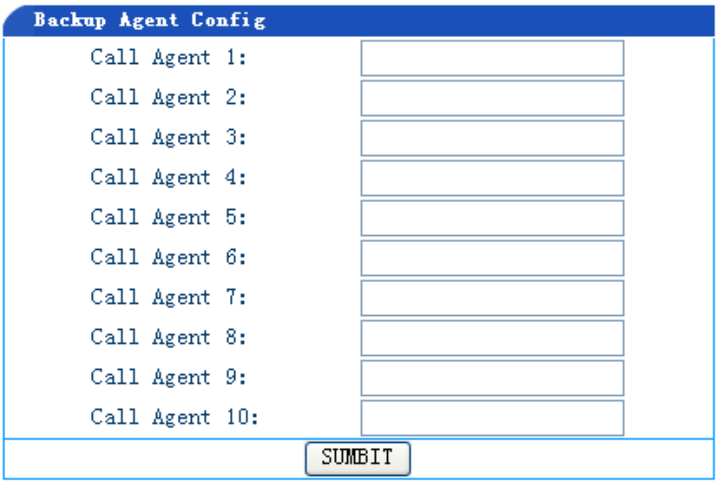


Figure 0-24 Backup Agent Config screen

Call Agent 1~10

Set call agent IP address and port number. No default value.

- Use ":" between IP address and port number.

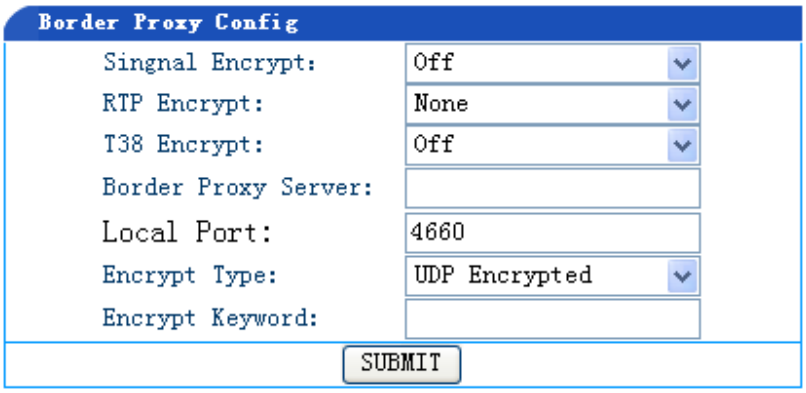
- Address can be either IP address or domain name. If using domain name, the user must set DNS server parameters in Network Settings page and enable the DNS service.
- Sample format of fully qualified addresses: 202.202.2.202:2727; callagent.com:2727.

Border Proxy Config

To connect to user agents, we also need to gather information about border agent, registration server, and local network domain name and IP address. Customer can be in different locations, and settings are all depending on their locations and local networks.

Click Advanced **Options > Border Proxy Config** from the left pane. The following displays:

Note: For information on how to use Submit, see most used buttons.



Border Proxy[26]

Set whether signaling and RTP are using a border agent. Default value is None. Possible settings are: None, Signaling, Signaling and RTP.

- **None:** Do not use border agent.
- **Signaling:** Only Signaling is going to use border agent.
- **Signaling and RTP:** Both Signaling and RTP stream are using border agent.

Border Proxy Server

[26] **Border Agent:** Also known as Border Controller, which normally includes Sign Proxy and Media Proxy function modules.

Set IP address and port number for border agent. No default value. Separate IP address and port number with a ":".

Local Port

Local port number for border agent. Default value is: 4660. Local port number can be anything, as long as it does not conflict with port numbers for other equipments.

Encrypt type

Set encryption method. Default value is: None.

Note: Encryption setting must be the same with what the border agent is using. Possible options are:

- **None:** TCP encryption, HTTPU mode. No encryption algorithm is used.
- **Encrypted:** TCP encryption, HTTPU mode. Encryption algorithm is used.
- **TCP Encrypted:** Encrypt signaling and RTP over TCP. Also use encryption algorithm.
- **TCP Not Encrypted:** Encrypt signaling and RTP over TCP, but no encryption algorithm is used.
- **UDP Not Encrypted:** Encrypt signaling and RTP over UDP, but no encryption algorithm is used.
- **UDP Encrypted:** UDP encryption. Also use encryption algorithm.
- **Using Keyword:** UDP encryption using backward keyword encryption algorithm
- **Using Keyword2:** UDP encryption using forward keyword encryption algorithm.
- **RC4**[27]: Using RC4 encryption algorithm.

Encrypt Keyword

Set encryption keyword when setting encryption method to "UDP Encrypted". Default value is: None.

- **Yes:** SIP Call ID will use server host name.
- **No:** SIP Call ID will use server IP address.

[27] **RC4**
The RC4 encryption algorithm is stream cipher, which can use variable length keys. The algorithm was developed by Ron Rivest, for RSA Data security. Analysis shows that the period of the cipher is overwhelmingly likely to be greater than 10^{100}. Eight to sixteen machine operations are required per output byte, and the cipher can be expected to run very quickly in software. Independent analysts have scrutinized the algorithm and it is considered secure.

EMS Optional

After log in, click **Advanced Config > EMS Config** from the left pane. The following displays:

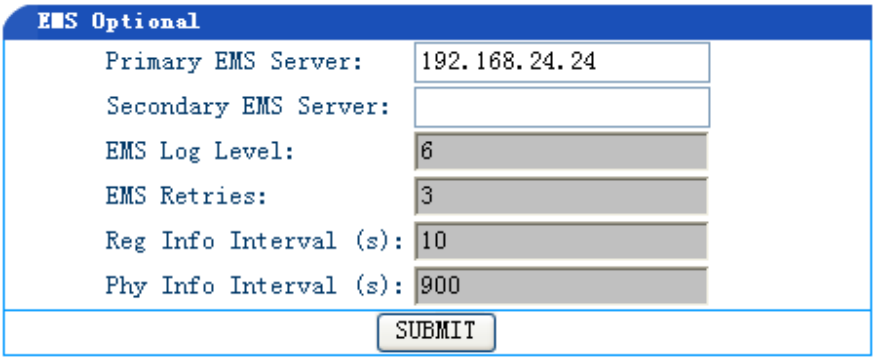


Figure 0-25 EMS Optional screen

Primary EMS Server

Enter primary EMS Server IP address if you want to use EMS service.

Secondary EMS Server

Enter secondary EMS server IP address.

EMS Log Level

This field value is automatically detected. You do not need to change this field.

EMS Retries

This field value is automatically detected. You do not need to change this field.

Reg Info Interval

This field value is automatically detected. You do not need to change this field.

Phy Info Interval

This field value is automatically detected. You do not need to change this field.

Bill Optional

After log in, click **Advanced Config > Bill Config** from the left pane. The following displays:

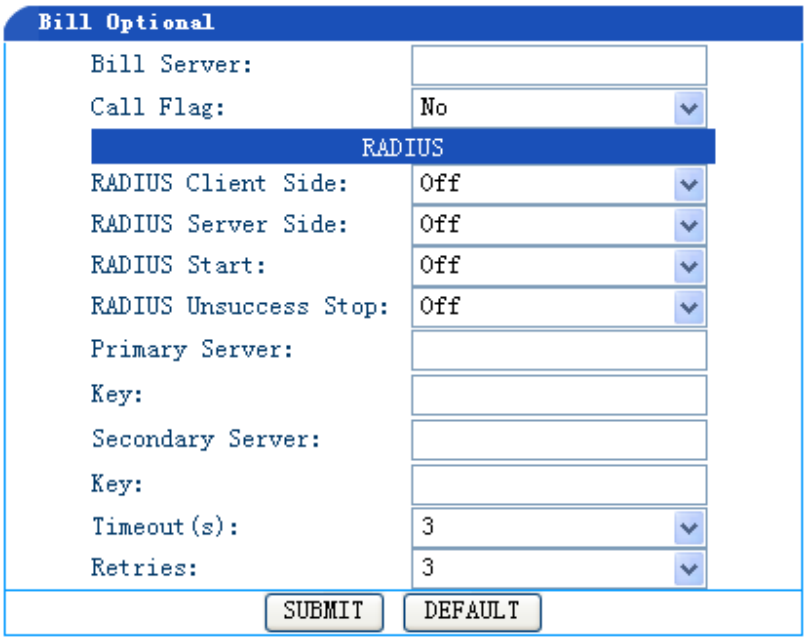


Figure 0-25 IMSOptional screen

Bill Server

In **Bill Server** field enter the IP address and the port of the bill server.

Call Flag

Select **Yes** or **No** to let the gateway work or not if the gateway lose communication with bill server.

RADIUS[28] Client side

Select **on** (to invoke) or **off** (not to invoke) to indicate to turn on or not the charging function of RADIUS client.

RADIUS Server side

[28] **RADIUS (Remote Authentication Dial In User Service)**
Remote Authentication Dial-In User Service (RADIUS) is a client/server protocol and software that enables remote access servers to communicate with a central server to authenticate dial-in users and authorize their access to the requested system or service. RADIUS allows a company to maintain user profiles in a central database that all remote servers can share. It provides better security, allowing a company to set up a policy that can be applied at a single administered network point. Having a central service also means that it's easier to track usage for billing and for keeping network statistics. Created by Livingston (now owned by Lucent), RADIUS is a de facto industry standard used by a number of network product companies and is a proposed IETF standard.

Select **on** (to invoke) or **off** (not to invoke) to indicate to turn on or not the charging function of RADIUS server.

RADIUS Start

Select **on** or **off** to indicate whether or not to transmit the initial RADIUS record when the charging function of RADIUS client or server is invoked.

RADIUS Unsuccess Stop

Select **on** or **off** to indicate whether or not to transmit RADIUS record of the unsuccessful calls when the charging function of RADIUS client or server is invoked.

Primary Server

In **Primary Server** field enter the IP address and the port of the primary RADIUS server. If no port is set, then the default port 1813 will be used.

Key

In **Key** field enter the share key for the communication between primary RADIUS client and server. Make sure the settings of both sides are consistent.

Secondary Server

In **Secondary Server** field enter the IP address and the port of the secondary RADIUS server. If no port is set, then the default port 1813 will be used.

Key

In **Key** field enter the share key for the communication between secondary RADIUS client and server. Make sure the settings of both sides are consistent.

Timeout

In **Timeout** field the default setting is 3 seconds.

Retries

In **Retries** field enter re-transmit times when the charging function is not responding. The default setting is 3 seconds.

Log Information

Call Status Information

Click **Log Information** link on the left of Figure 4-1. Then click Call Info. The Call Info page displays:

Figure 0-26 Call Log Info screen

status	off/on-hook and ringing status.
ts	timeslot.
d	DSP. This field shows the DSP chip used.
c	channel. This field indicates the channel of DSP.
call	identify one call number，which is a random number.
remote	remote IP address followed by RTP port number.
local	the local RTP port number.
codec	encoding and decoding. GT8 support the following codec: G729A/20,iLBC/30,G723/30,GSM/20,PCMU/20,PCMA/20
state	call state, which indicates the current state. It can be SEND; DELIVERED; PRESENT; RECEIVED, and ACTIVE.
number	phone number. (C)：calling number；(D)：called number.
timestamp	which has two types：one is setup time，whose duration is 0； another is connection time. In the information shown on screen, the former is setup time, while the latter is connection time, the unit is in seconds.
caller id	this is a length of digit used to identify a call when SIP is switching information; the length and value of the digit are randomly generated.

Resources Information

Click **Log Info** link on the left of Figure 4-1. Then click **Resource Info**. The Resource Info page displays. In this page, you can see the logon information (including the IP address and level of the logon user) of all WEB users, SIP registration information, telephone information and RTP information.

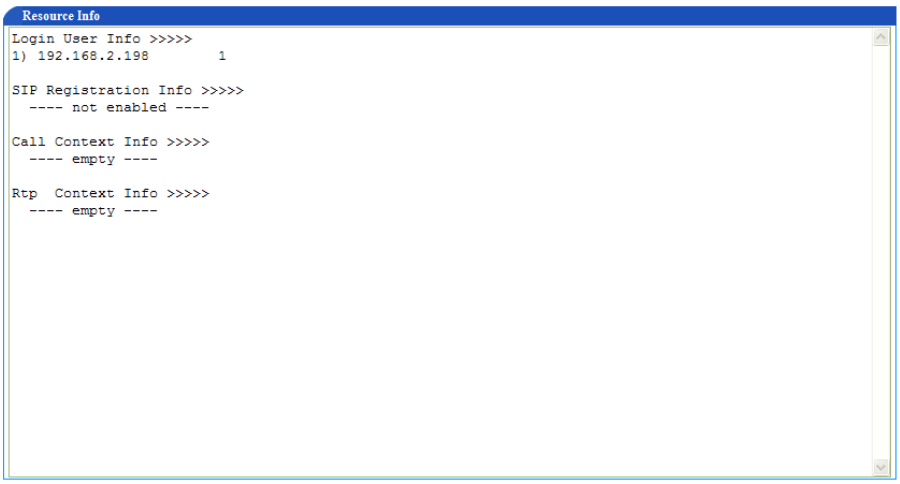


Figure 0-27 Resource Info screen

Message Information

Click **Log Info** link on the left of Figure 4-1. The click Message Log. The Log Info page displays. You can check all the call related information in this page.

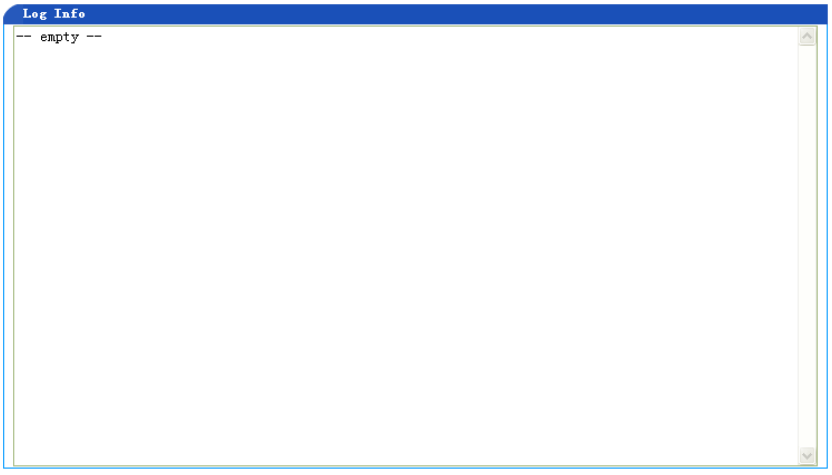


Figure 0-28 Log Info screen

Error Information

Click **Log Info** link on the left of Figure 4-1. Then click **Error Log**. The Log Info page displays. You can check all the errors, logons, exits and overtime web access information in this page.

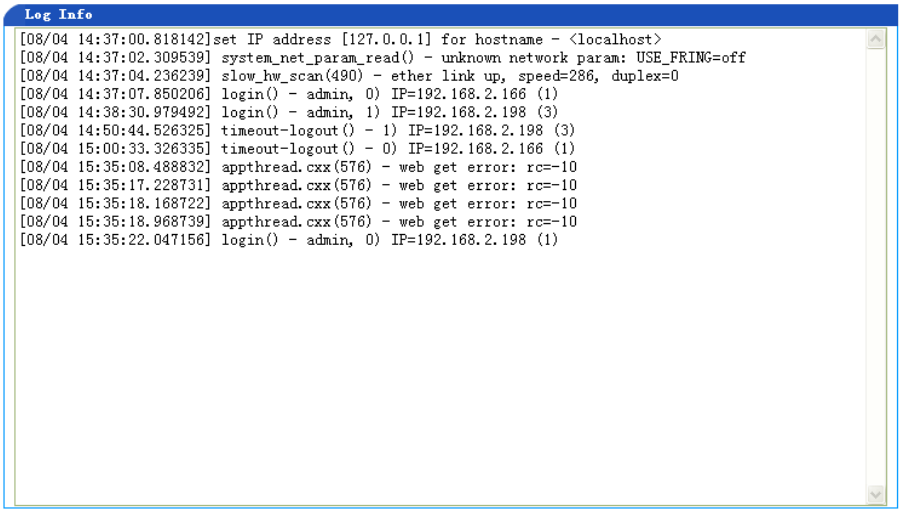


Figure 0-29 Error Info screen

Startup Information

Click **Log Info** link on the left of Figure 4-1. Then click **Startup Info**. The Log Info page displays. You can check all the startup information of the gateway from this page.

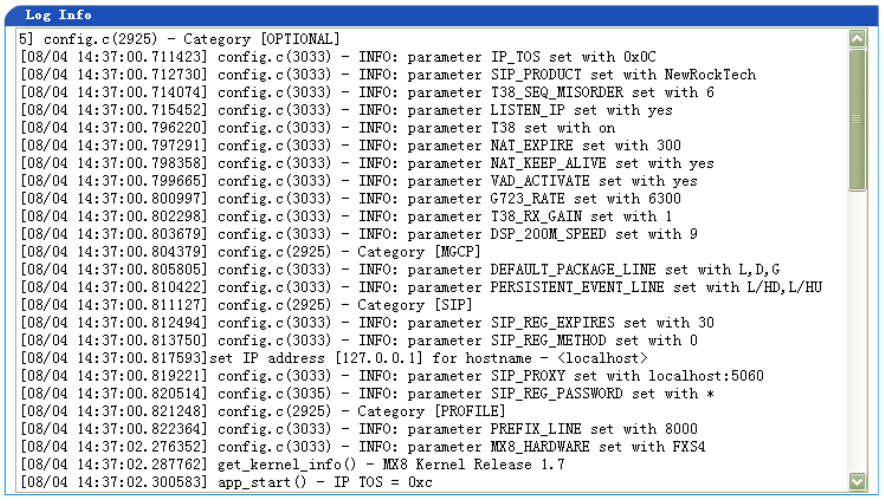

Log Info
5] config.c(2925) - Category [OPTIONAL]
[08/04 14:37:00.711423] config.c(3033) - INFO: parameter IP_TOS set with 0x0C
[08/04 14:37:00.712730] config.c(3033) - INFO: parameter SIP_PRODUCT set with NewRockTech
[08/04 14:37:00.714074] config.c(3033) - INFO: parameter T38_SEQ_MISORDER set with 6
[08/04 14:37:00.715452] config.c(3033) - INFO: parameter LISTEN_IP set with yes
[08/04 14:37:00.796220] config.c(3033) - INFO: parameter T38 set with on
[08/04 14:37:00.797291] config.c(3033) - INFO: parameter NAT_EXPIRE set with 300
[08/04 14:37:00.798358] config.c(3033) - INFO: parameter NAT_KEEP_ALIVE set with yes
[08/04 14:37:00.799665] config.c(3033) - INFO: parameter VAD_ACTIVATE set with yes
[08/04 14:37:00.800997] config.c(3033) - INFO: parameter G723_RATE set with 6300
[08/04 14:37:00.802298] config.c(3033) - INFO: parameter T38_RX_GAIN set with 1
[08/04 14:37:00.803679] config.c(3033) - INFO: parameter DSP_200M_SPEED set with 9
[08/04 14:37:00.804379] config.c(2925) - Category [MGCP]
[08/04 14:37:00.805805] config.c(3033) - INFO: parameter DEFAULT_PACKAGE_LINE set with L,D,G
[08/04 14:37:00.810422] config.c(3033) - INFO: parameter PERSISTENT_EVENT_LINE set with L/HD,L/HU
[08/04 14:37:00.811127] config.c(2925) - Category [SIP]
[08/04 14:37:00.812494] config.c(3033) - INFO: parameter SIP_REG_EXPIRES set with 30
[08/04 14:37:00.813750] config.c(3033) - INFO: parameter SIP_REG_METHOD set with 0
[08/04 14:37:00.817593]set IP address [127.0.0.1] for hostname - <localhost>
[08/04 14:37:00.819221] config.c(3033) - INFO: parameter SIP_PROXY set with localhost:5060
[08/04 14:37:00.820514] config.c(3035) - INFO: parameter SIP_REG_PASSWORD set with *
[08/04 14:37:00.821248] config.c(2925) - Category [PROFILE]
[08/04 14:37:00.822364] config.c(3033) - INFO: parameter PREFIX_LINE set with 8000
[08/04 14:37:02.276352] config.c(3033) - INFO: parameter MX8_HARDWARE set with FXS4
[08/04 14:37:02.287762] get_kernel_info() - MX8 Kernel Release 1.7
[08/04 14:37:02.300583] app_start() - IP TOS = 0xc

Figure 0-30 Startup Info screen

Clear Message Information

Click **Log Info** link on the left of Figure 4-1. Then click **Clear Msg Log**. Now you can clear the information in Message Information.

System Tools

Restore Factory Setting

Click **System Tools** on the left of Figure 4-1. Then click Restore Factory Setting. The following displays.

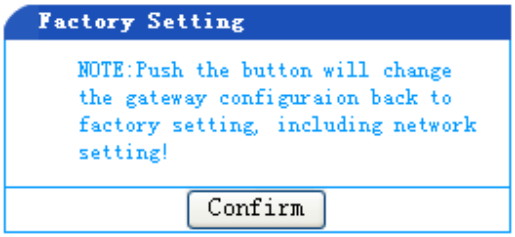

Factory Setting
NOTE:Push the button will change the gateway configuraion back to factory setting, including network setting!
Confirm

Figure 0-31 Restore Factory setting screen

Click the Confirm button to restore default factory settings.

GT8 gateway has most parameters set to commonly used default value. In most cases, customers do not have to set the parameters for themselves. See Index for details about factory default settings.

Software Update

Click **System Tools** on the left of Figure 4-1. Then click **Software Update**. The following displays:

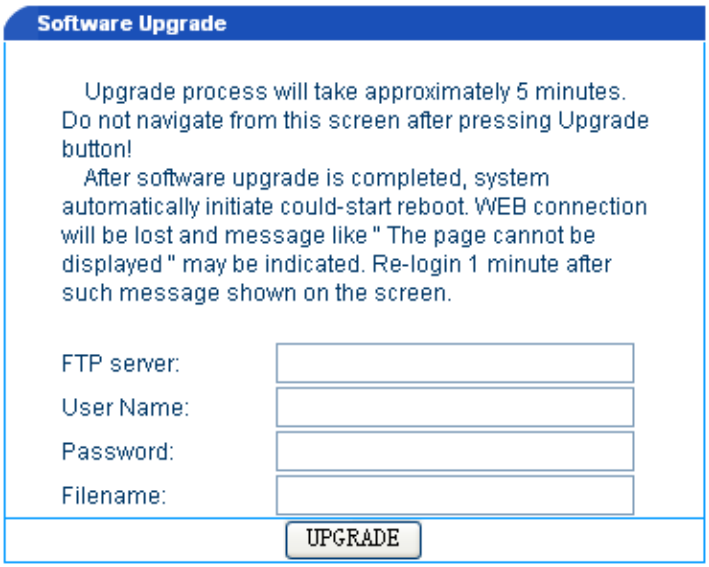

Software Upgrade
Upgrade process will take approximately 5 minutes. Do not navigate from this screen after pressing Upgrade button!
After software upgrade is completed, system automatically initiate could-start reboot. WEB connection will be lost and message like " The page cannot be displayed " may be indicated. Re-login 1 minute after such message shown on the screen.
FTP server:
User Name:
Password:
Filename:
UPGRADE

Figure 0-32 Software Update screen

FTP Server

Please enter the IP address or domain name of the FTP server which is used to update the software.

Filename

Please enter the filename of the software version which you need to update; otherwise, the software will be updated to the latest version.

Note： No operation is permitted during update period! The “Reboot” message will pop up when the update is successful. Click the OK button, the software will switch to “Reboot” page automatically. After clicking the Reboot button, please turn off power manually and restart the gateway.

Change Password

Click **System Tools** on the left of Figure 4-1. Then click **Change Password**. The following displays:

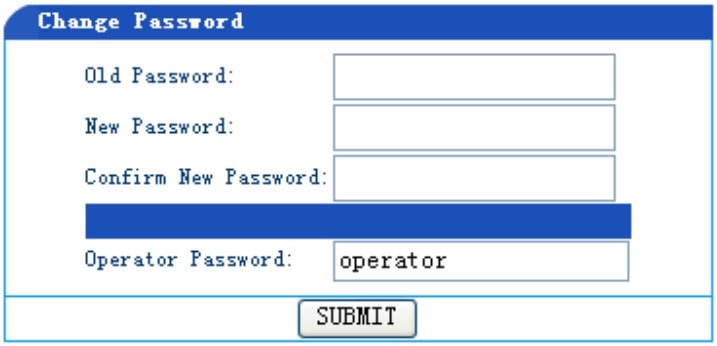


Figure 0-33 Change Password screen

Only an administrator has the authority to changes password. The first three fields are used to change administrator password. Please input the old password in the **Old password** field, and input the new password in the **New password** field. Enter the new password again in **Confirm new password** field; then click the Submit button to finish.

The current operator password is displayed in plain text mode. An administrator can change it at any time and does not need to input the current administrator password when he/she wants to change the operator password. Just enter a new password in **Operator Password** field, and click the Submit button to finish.

Restart Gateway

Click **System Tools** on the left of Figure 4-1. Then click **Reboot**. After that, click the REBOOT button to restart the gateway.

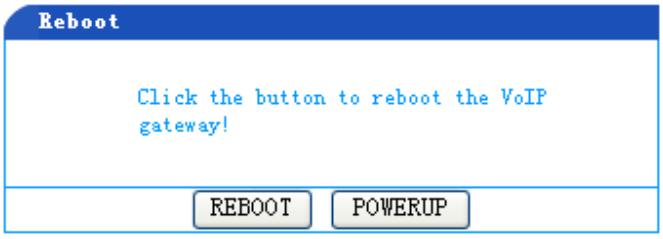


Figure 0-34 Restart Gateway screen

Exit

Click **Logout** on the left of Figure 4-1. Then exit WEB operation, and display the login page.

6

APPENDIX

Factory Default Settings

Default System Parameter Settings

RTP Port Min：10010
RTP Port Max：10030
1st Digit Timeout (Second)：12
Inter Digit Timeout (Second)：12
Critical Digit Timeout (Second)：5
DTMF Transmit Mode：AUDIO
Default Codec：PCMU/20, G729A/20, iLBC/30, G723/30, GSM/20, PCMA/20
Echo Cancellation：on

Default MGCP Settings

MGCP Port：2427
Default Package：L, D, G
Persistent Line Event：L/HD, L/HU, L/HF
Wildcard：no
All Wildcard：no
End-of-Line Using CR：no
Quarantine Default to Loop：no
Default Package Don't Send Name：no
Always Enable 1st Dgt Timeout：no
On-hook don't Delete Connection：no
Notify Instead of 401/402：no
Using L Package Handle FXO：yes
Using Configured Digit Map：no

Default SIP Settings

SIP Port：5060
Registration Expires (Second)：30
Authentication Mode：Per Endpoint

Default Network Parameter Settings

Gateway IP Address：192.168.2.1
DHCP：on
Local IP Address：192.168.2.218
Subnet Mask：255.255.255.0
DNS：off
PPPoE：off
Preference TIME Server：192.43.244.18
Alternate TIME Server：198.60.22.240
Timeout (Minute)：10
Query Interval (Minute)：120

Default FXS Settings

Forbid to Call：off
Call Waiting：off
Call Holding：off
Call Forward：off
Caller ID：off
CID on Call Waiting：off
Anonymous Call：off
Hotline：off
Delay Hotline：off
No Disturb：off
Speed Dial：off
Fashion Ring：off
Reverse Battery：off

Default FXO Settings

Forbid to Call：off
Hotline：off
Dial Ton：on
Echo Cancellation：on
Detect FSK：off

System Advanced Default Settings

Event Log Type：FILE
Event Log Level：3
Country ID：China
Forwarding Number Mode：Forwarding Number

Fashion Ring Max：20
SNMP Port：2700
SNMP Trap Port：162
NAT Keep Alive：no
STUN：off
RADIUS Client Side：off
RADIUS Server Side：off
RADIUS Start：off
RADIUS Unsuccess Stop：off
Timeout (Second)：3
Retries：3

FXO Advanced Default Settings

FXO Gain：-3.5
FXO Relay Time (ms)：400
Digit on Time (ms)：100
Digit off Time (ms)：100
Buy Tone Repetition：2
Busy Tone Parameter1：61485
Busy Tone Parameter2：0
Busy Tone on Time (ms)：350
Busy Tone off Time (ms)：350

FXS Advanced Default Settings

FXS Gain：-7.0
Hookflash Min (ms)：75
Hookflash Max (ms)：800
Hook Status Changes (ms)：50

Default IP Advanced Settings

RTP Jitter Max (Frame)：50
RTP Jitter Min (Frame)：3
2833 Packet Type：97
Send Busy Tone for Network Breakdown：off
Switch to FXO for Network Breakdown：on
Generation of the Mute Compress and Comfort Noise：yes
G.723.1 Rate (BPS)：5300
IP Packet in Tos Field：0x0C
T.38：on
T.38 Data Frame Length (ms)：40
T.38 Redundancy Fram Number：4
T.38 Change UDP Port：no
T.38 Error Detect Mode：off

SIP Advanced Default Settings

Response Default Port Using Received Port：no
Response Default Port Using Proxy Port：no
RTP Port Mapping：no
Replace 18x with 180：no
CPN from Request Line：no
Response Do Not Check Via：yes
Use the Full Domain Name Information during Registering：yes
Keep the Original Contact Information：no
SIP VIA using NAT Information：yes
SIP TO adopt Domain Name Information：yes

Glossary

DHCP（Dynamic Host Configuration Protocol）

DHCP (Dynamic Host Configuration Protocol) is a network protocol used to assign TCP/IP addresses to client servers. Each client server is connected to the central DHCP server, which gives the network configuration of each client, including the IP address, gateway and DNS server information.

DSP（Digital Signal Processing）

Adjust, and filtrate digital frequency.

RTP（Real-Time Transport Protocol）

RTP is an Internet protocol standard that specifies a way for programs to manage the real-time transmission of multimedia data over either unicast or multicast network services. RTP is defined as working one to one or one to more, which can provide real time. RTP usually using UDP (User Datagram Protocol) to transfer data, but RTP also works for TCP (Transmission Control Protocol) or ATM (Asynchronous Transfer Mode).There are 2 ports when a program starts a RTP communication: one for RTP and one for RTCP. RTP does not address resource reservation and does not guarantee quality-of-service for real-time services. The data transport is augmented by a control protocol (RTCP) to allow monitoring of the

data delivery in a manner scalable to large multicast networks, and to provide minimal control and identification functionality. RTP and RTCP are designed to be independent of the underlying transport and network layers. The protocol supports the use of RTP-level translators and mixers.

RTP port refers to sending and receiving port.

RTP provides support for media data packetization and real time transmission. Every RTP packet consists of a Header and a Payload. The first 12 bits are RTP Fixed Header Fields. Payload can be either video or audio. Figure 0-1 shows the RTP header format.

Figure 0-1: RTP Header Format

Key header fields and their meanings:

- **CSRC Count (CC)**: 4 bits. The CSRC count contains the number of CSRC identifiers that follow the fixed header. For example, one CSRC list can represent a audio conference. This call uses a RTP mixer to combine audios of all callers into an RTP data stream.
- **Payload Type (PT):** 7 bits. Indicates payload format, including codec, clock rate, channel, etc. For example, type 2 indicates payload in this packet is using ITU G721 codec, sample rate is 8000Hz and using single channel.
- **Sequence Number:** 16 bits. The sequence number is mainly used to detect losses. RTP does not try to re-transmit for detected losses. It's up the application to handle lost packets.
- **Time Stamp:** 32 bits. The timestamp is used to place the incoming audio and video packets in the correct timing order (playout delay compensation). The sequence number is mainly used to detect losses. Sequence numbers increase by one for each RTP packet transmitted, timestamps increase by the time "covered" by a packet. For video formats where a video frame is split across several RTP packets, several packets may have the same timestamp. In some cases such as carrying DTMF (touch tone) data (RFC 2833), RTP timestamps may not be monotonic.

DTMF（Dual Tone Multi-Frequency）

In PSTN service, after a call is connected, user's touch tone info is transmitted via DTMF, also known as second dial tone information. It is widely used in intelligent network and value-added services.

- Audio: Voice data transparent transmit mode.
- 2833: A special RTP packet. PT field of the header indicates this is a DTMF packet. See FTC 2833 for details.
- INFO: Information transmission mode. Optional way of DTMF transmission. As in SIP messages, use INFO to indicate a DTMF signal.

Speech CODEC

Also called a "voice codec" or "vocoder," it is a hardware circuit that converts the spoken word into digital code and vice versa. It comprises the A/D and D/A conversion and compression technique. If music is encoded with a speech codec, it will not sound as good when decoded at the other end. A speech codec is an audio codec designed for human voice. By analyzing vocal tract sounds, a recipe for rebuilding the sound at the other end is sent rather than the soundwaves themselves. The speech codec is able to achieve a much higher compression ratio, which results in a smaller amount of digital data for transmission. When telephones were first digitized in the early 1960s, they generated digital streams of 64 Kbps. Since then, speech CODECS have reduced voice to as little as 5 Kbps and less.

Echo Cancellation

The term echo cancellation is used in telephony to describe the process of removing echo from a voice communication in order to improve voice quality on a telephone call. In addition to improving quality, this process improves bandwidth savings achieved through silence suppression by preventing echo from traveling across a network.

There are two types of echo of relevance in telephony: acoustic echo and hybrid echo. Speech compression techniques and digital processing delay often contribute to echo generation in telephone networks. Echo cancellation involves first recognizing the originally transmitted signal that re-appears, with some delay, in the transmitted or received signal. Once the echo is recognized, it can be removed by 'subtracting' it from the transmitted or received signal.

This technique is generally implemented using a digital signal processor (DSP), but can also be implemented in software. Echo

cancellation is done using either echo suppressors or echo cancellers.

MGCP (Media Gateway Control Protocol)

Media Gateway Control Protocol (MGCP) is used for controlling telephony gateways from external call control elements called media gateway controllers or call agents. A telephony gateway is a network element that provides conversion between the audio signals carried on telephone circuits and data packets carried over the Internet or over other packet networks.

MGCP assumes a call control architecture where the call control intelligence is outside the gateways and handled by external call control elements. The MGCP assumes that these call control elements, or Call Agents, will synchronize with each other to send coherent commands to the gateways under their control. MGCP is, in essence, a master/slave protocol, where the gateways are expected to execute commands sent by the Call Agents.

The MGCP implements the media gateway control interface as a set of transactions. The transactions are composed of a command and a mandatory response. There are nine types of commands:

MGCP Commands (MGC=Media Gateway Controller; MG=Media Gateway)

MGC --> MG CreateConnection: Creates a connection between two endpoints; uses SDP to define the receive capabilities of the participating endpoints.

MGC --> MG ModifyConnection: Modifies the properties of a connection; has nearly the same parameters as the CreateConnection command.

MGC <--> MG DeleteConnection: Terminates a connection and collects statistics on the execution of the connection.

MGC --> MG NotificationRequest: Requests the media gateway to send notifications on the occurrence of specified events in an endpoint.

MGC <-- MG Notify: Informs the media gateway controller when observed events occur.

MGC --> MG AuditEndpoint: Determines the status of an endpoint.

MGC --> MG AuditConnection: Retrieves the parameters related to a connection.

MGC <-- MG RestartInProgress: Signals that an endpoint or group of endpoints is taking in or out of service.

MGC --> MG: Endpoint Configuration

The first four commands are sent by the Call Agent to a gateway. The Notify command is sent by the gateway to the Call Agent. The gateway may also send a DeleteConnection. The Call Agent may send either of the Audit commands to the gateway. The Gateway may send a RestartInProgress command to the Call Agent.

All commands are composed of a command header, optionally followed by a session description. All responses are composed of a response header, optionally followed by a session description. Headers and session descriptions are encoded as a set of text lines, separated by a carriage return and line feed character (or, optionally, a single line-feed character). The headers are separated from the session description by an empty line.

MGCP uses a transaction identifier to correlate commands and responses. Transaction identifiers have values between 1 and 999999999. An MGCP entity cannot reuse a transaction identifier sooner than 3 minutes after completion of the previous command in which the identifier was used.

The command header is composed of:

- A command line, identifying the requested action or verb, the transaction identifier, the endpoint towards which the action is requested, and the MGCP protocol version,
- A set of parameter lines, composed of a parameter name followed by a parameter value.

The command line is composed of:

- Name of the requested verb.
- Transaction identifier correlates commands and responses. Values may be between 1 and 999999999. An MGCP entity cannot reuse a transaction identifier sooner than 3 minutes after completion of the previous command in which the identifier was used.
- Name of the endpoint that should execute the command (in notifications, the name of the endpoint that is issuing the notification).
- Protocol version.

These four items are encoded as strings of printable ASCII characters, separated by white spaces, i.e., the ASCII space (0x20) or tabulation (0x09) characters. It is recommended to use exactly one ASCII space separator.

MGCP Call Agent

Call Agent, also known as Media Gateway Controller, controls the Media Gateway. In MGCP, a call agent primarily handles all the call processing by linking with the IP network through constant communications with an IP signaling device, for example an SIP Server or an H.323 gatekeeper.

Call Agent is comprised of the call control "intelligence" and a media gateway boasting the media functions, for example conversion from TDM voice to Voice over IP.

Media Gateways feature endpoints for the Call Agent to create and manage media sessions with other multimedia endpoints. Endpoints are sources and/or sinks of data that can be physical or virtual. For creating physical endpoints, hardware installation is needed while virtual endpoint can be created using available software.

Call Agents come with the capability of creating new connections, or modify an existing connection. Generally, a media gateway is a network element which provides conversion between the data packets carried over the Internet or other packet networks and the voice signals carried by telephone lines. The Call Agent provides instructions to the endpoints to check for any events and - if there is any - create signals. The endpoints are designed in such a way as to automatically communicate changes in service state to the Call Agent. The Call Agent can audit endpoints and the connections on endpoints.

401/402 Response Code

Response Code is a 3-digit response to the request, indicating the processing results for requests. For example, 401 and 402 represent responses to the on-hook and off-hook operations.

NTFY

Notification, or Notify, a command sent from gateway to call agent.

SIP (Session Initiation Protocol)

Session Initiation Protocol (SIP) is the Internet Engineering Task Force's (IETF's) standard for multimedia conferencing over IP. SIP is an ASCII-based, application-layer control protocol (defined in RFC 2543) that can be used to establish, maintain, and terminate calls between two or more end points.

Like other VoIP protocols, SIP is designed to address the functions of signaling and session management within a packet telephony

network. Signaling allows call information to be carried across network boundaries. Session management provides the ability to control the attributes of an end-to-end call.

SIP provides the capabilities to:

- Determine the location of the target end point—SIP supports address resolution, name mapping, and call redirection.
- Determine the media capabilities of the target end point—Via Session Description Protocol (SDP), SIP determines the "lowest level" of common services between the end points. Conferences are established using only the media capabilities that can be supported by all end points.
- Determine the availability of the target end point—If a call cannot be completed because the target end point is unavailable, SIP determines whether the called party is already on the phone or did not answer in the allotted number of rings. It then returns a message indicating why the target end point was unavailable.
- Establish a session between the originating and target end point—If the call can be completed, SIP establishes a session between the end points. SIP also supports mid-call changes, such as the addition of another end point to the conference or the changing of a media characteristic or codec.
- Handle the transfer and termination of calls—SIP supports the transfer of calls from one end point to another. During a call transfer, SIP simply establishes a session between the transferee and a new end point (specified by the transferring party) and terminates the session between the transferee and the transferring party. At the end of a call, SIP terminates the sessions between all parties.

  Conferences can consist of two or more users and can be established using multicast or multiple unicast sessions.

Components of SIP

SIP is a peer-to-peer protocol. The peers in a session are called User Agents (UAs). A user agent can function in one of the following roles:

- User agent client (UAC)—A client application that initiates the SIP request.
- User agent server (UAS)—A server application that contacts the user when a SIP request is received and that returns a response on behalf of the user.
- Typically, a SIP end point is capable of functioning as both a UAC and a UAS, but functions only as one or the other per transaction. Whether the endpoint functions as a

UAC or a UAS depends on the UA that initiated the request.

- From an architecture standpoint, the physical components of a SIP network can be grouped into two categories: clients and servers. Figure 1-1 illustrates the architecture of a SIP network.
- SIP Clients
- SIP clients include:
    - Phones - Can act as either a UAS or UAC. Softphones (PCs that have phone capabilities installed) and Cisco SIP IP phones can initiate SIP requests and respond to requests.
    - Gateways - Provide call control. Gateways provide many services, the most common being a translation function between SIP conferencing endpoints and other terminal types. This function includes translation between transmission formats and between communications procedures. In addition, the gateway translates between audio and video codecs and performs call setup and clearing on both the LAN side and the switched-circuit network side.
    - SIP Servers

SIP servers include:

- Proxy server - The proxy server is an intermediate device that receives SIP requests from a client and then forwards the requests on the client's behalf. Basically, proxy servers receive SIP messages and forward them to the next SIP server in the network. Proxy servers can provide functions such as authentication, authorization, network access control, routing, reliable request retransmission, and security.
- Redirect server - Provides the client with information about the next hop or hops that a message should take and then the client contacts the next hop server or UAS directly.
- Registrar server - Processes requests from UACs for registration of their current location. Registrar servers are often co-located with a redirect or proxy server.

How SIP Works

SIP is a simple, ASCII-based protocol that uses requests and responses to establish communication among the various components in the network and to ultimately establish a conference between two or more end points.

Users in a SIP network are identified by unique SIP addresses. A SIP address is similar to an e-mail address and is in the format of sip:userID@gateway.com. The user ID can be either a user name or an E.164 address.

Users register with a registrar server using their assigned SIP addresses. The registrar server provides this information to the location server upon request.

When a user initiates a call, a SIP request is sent to a SIP server (either a proxy or a redirect server). The request includes the address of the caller (in the From header field) and the address of the intended callee (in the To header field). The following sections provide simple examples of successful, point-to-point calls established using a proxy and a redirect server.

Over time, a SIP end user might move between end systems. The location of the end user can be dynamically registered with the SIP server. The location server can use one or more protocols (including finger, rwhois, and LDAP) to locate the end user. Because the end user can be logged in at more than one station and because the location server can sometimes have inaccurate information, it might return more than one address for the end user. If the request is coming through a SIP proxy server, the proxy server will try each of the returned addresses until it locates the end user. If the request is coming through a SIP redirect server, the redirect server forwards all the addresses to the caller in the Contact header field of the invitation response.

For more information, see RFC 2543—SIP: Session Initiation Protocol, which can be found at http://www.faqs.org/rfcs/.

Using a Proxy Server

If a proxy server is used, the caller UA sends an INVITE request to the proxy server, the proxy server determines the path, and then forwards the request to the callee.

The callee responds to the proxy server, which in turn, forwards the response to the caller.

The proxy server forwards the acknowledgments of both parties. A session is then established between the caller and callee. Real-time

Transfer Protocol (RTP) is used for the communication between the caller and the callee.

Using a Redirect Server

If a redirect server is used, the caller UA sends an INVITE request to the redirect server, the redirect server contacts the location server to determine the path to the callee, and then the redirect server sends that information back to the caller. The caller then acknowledges receipt of the information.

The caller then sends a request to the device indicated in the redirection information (which could be the callee or another server that will forward the request). Once the request reaches the callee, it sends back a response and the caller acknowledges the response. RTP is used for the communication between the caller and the callee.

SIP Versus H.323

In addition to SIP, there are other protocols that facilitate voice transmission over IP. One such protocol is H.323. H.323 originated as an International Telecommunications Union (ITU) multimedia standard and is used for both packet telephony and video streaming. The H.323 standard incorporates multiple protocols, including Q.931 for signaling, H.245 for negotiation, and Registration Admission and Status (RAS) for session control. H.323 was the first standard for call control for VoIP and is supported on all Cisco Systems' voice gateways.

SIP and H.323 were designed to address session control and signaling functions in distributed call control architecture. Although SIP and H.323 can also be used to communicate to limited intelligence end points, they are especially well-suited for communication with intelligent end points.

Although SIP messages are not directly compatible with H.323, both protocols can coexist in the same packet telephony network if a device that supports the interoperability is available.

For example, a call agent could use H.323 to communicate with gateways and use SIP for inter-call agent signaling. Then, after the bearer connection is set up, the bearer information flows between the different gateways as an RTP stream..

Proxy

Proxy is the kernel of SIP, implementing message transfer functions.

Registrar

When a client powers on, it will tell network its IP address in order to be found. We call this procedure “register”. The server that accepts this request is called “registrar”.

Registration Expire(s)

In order to control client side, every register message has a certain stored period. If the message is modified in that period, which mean it works for user otherwise Registrar will consider the message is not useful any more, so it will be deleted.

DNS (Domain Name System, or Service or Server）

DNS is a very important service of internet, an Internet service that translates domain names into IP addresses. Because domain names are alphabetic, they're easier to remember. The Internet however, is really based on IP addresses. Every time you use a domain name, therefore, a DNS service must translate the name into the corresponding IP address. For example, the domain name www.example.com might translate to 198.105.232.4.

PPPoE（Point-to-Point Protocol Over Ethernet）

PPPoE relies on two widely accepted standards: PPP and Ethernet. PPPoE is a specification for connecting the users on an Ethernet to the Internet through a common broadband medium, such as a single DSL line, wireless device or cable modem. The feature of PPPoE:

- All the users over the Ethernet share a common connection
- Allow single user P2P to different network
- Ethernet principles supporting multiple users in a LAN combine with the principles of PPP, which apply to serial connections.

Time Server

Time Server provides time calibration, control, and synchronization for equipments running in the network.

Caller ID Detecting

Using FXO to detect caller ID from the PSTN and use this number as the originating number for the IP call.

SNMP (Simple Network Management Protocol)

The Simple Network Management Protocol (SNMP) is an application layer protocol that facilitates the exchange of management information between network devices. It is part of the Transmission Control Protocol/Internet Protocol (TCP/IP) protocol suite. SNMP enables network administrators to manage network performance, find and solve network problems, and plan for network growth.

Two versions of SNMP exist: SNMP version 1 (SNMPv1) and SNMP version 2 (SNMPv2). Both versions have a number of features in common, but SNMPv2 offers enhancements, such as additional protocol operations. Standardization of yet another version of SNMP—SNMP Version 3 (SNMPv3) - is pending.

SNMP basic components:

SHORT NAME	DESCRIPTION
MIB	MANAGEMENT INFORMATION BASES
SMI	STRUCTURE OF MANAGEMENT INFORMATION
SNMP	SIMPLE NETWORK MANAGEMENT PROTOCOL

There are two ways to get data from managed network equipment, one is polling-only, which engages the workstation all the time; the other is interrupt-based method, which gets data periodically. Both methods have their disadvantages. The result is the combined method, called trap-directed polling. This is probably the most effective way to gather managed network data.

UDP Port

Port has two meanings in network technology. One is the physical port, via which other equipments can be connected. The other is the logical port, interfacing with different network protocols. Logical port can be of two types as well, depending what protocols we are talking about. One is TCP port, the other is UDP port. UDP port is a protocol port for data packages. No connection between client and server is required to use this port.

SNMP Trap

Trap is part of SNMP. A trap is a one-way message from a network element, such as a router, switch or server, to the NMS. The messages are sent via UDP, which means they are not guaranteed to arrive.

NAT（Network Address Translator or Translation）

Network Address Translation, an Internet standard that enables a local-area network (LAN) to use one set of IP addresses for internal traffic and a second set of addresses for external traffic. A NAT box located where the LAN meets the Internet makes all necessary IP address translations.

SDP (Session Description Protocol)

SDP describes multimedia sessions for the purpose of session announcement, session invitation and other forms of multimedia session initiation.

STUN（Simple Traversal of UDP over NATs）

A protocol that allows applications to detect that a network address translation (NAT) is being used. It can also detect the type of NAT and IP address assigned by it. STUN was developed to support interactive, two-way communications over the Internet such as for

voice (VoIP) and videoconferencing. The STUN client sends requests to a STUN server, which is typically hosted by the service provider.

Unlike application layer gateways (ALGs) and Middlebox Communications (MIDCOM), which also support two-way communications through NATs, STUN requires no changes to the NAT.

RADIUS（Remote Authentication Dial In User Service）

Remote Authentication Dial-In User Service (RADIUS) is a client/server protocol and software that enables remote access servers to communicate with a central server to authenticate dial-in users and authorize their access to the requested system or service. RADIUS allows a company to maintain user profiles in a central database that all remote servers can share. It provides better security, allowing a company to set up a policy that can be applied at a single administered network point. Having a central service also means that it's easier to track usage for billing and for keeping network statistics. Created by Livingston (now owned by Lucent), RADIUS is a de facto industry standard used by a number of network product companies and is a proposed IETF standard.

RADIUS Server

RADIUS Server authenticates clients and returns corresponding settings.

Signal Gain

When detected signals are not strong enough over the network, we use signal gain parameter to increase the strength of the signal.

Line Impedance

A measure of the total opposition to current flow in an alternating current circuit, made up of two components, ohmic resistance and reactance, and usually represented in complex notation as Z = R + iX, where R is the ohmic resistance and X is the reactance. It also refers

to an analogous measure of resistance to an alternating effect, as the resistance to vibration of the medium in sound transmission.

Signal Mode

Signal mode refers to inserting a silence signal periodically into the voice stream. Standard used in China specifies “450 Hz, 350ms ON + 350ms OFF”, means signal cycle is 700ms; insert 350 ms signal plus a silence signal of 350 ms for each cycle.

Jitter Buffer

Jitter is a major factor affecting the quality of IP calls. Jitter buffer is a software process that eliminates jitter caused by transmission delays in Internet telephony (VoIP) network. As the jitter buffer receives voice packets, it adds small amounts of delay to the packets so that all of the packets appear to have been received without delays. Voice signals are sequential by nature (i.e., they must be played back in the order in which they were sent) and the jitter buffer ensures that the received packets are in the correct order. Without a jitter buffer to smooth the transmission, data can be lost, resulting in choppy audio signals. There are two types of jitter buffers - dynamic and static. A static jitter buffer is hardware-based and configured by the manufacturer. A software-based jitter buffer is called a dynamic jitter buffer and can be configured by the system or network administrator.

RTP Payload Type

RTP Payload type indicates payload format, including codec, clock rate, channel, etc. For example, type 2 indicates payload in this packet is using ITU G721 codec, sample rate is 8000Hz and using single channel.

SID (Silence Information Description)

SID describes parameters to apply silence compression and comfort noise generation.

Voice Proxy

This is primarily used to resolve problems of accessing networks (for example, ADSL, Cable Modem, Ethernet, Proxy, etc.) using a faux/hidden IP or internal IP. It enables endpoints to access the network without having to change network settings. For IAD using public network and real IP address, voice proxy is not needed.

Symmetric RTP

RTP receiving end and sending end are using the same port.

Kernel

Kernel refers to the Linux Operating System. GED gateway uses Linux OS.

SDP（Session Description Protocol）

SDP describes multimedia sessions for the purpose of session announcement, session invitation and other forms of multimedia session initiation.

G.723.1 Voice CODEC

G.723.1 dual rate speech coder performs compression and decompression of 8 kHz speech signals. It encodes 16-bit PCM samples into 16-bit code-words yielding 10 or 12 code-words per 240 sample frames for the 5.3 Kbps and 6.3 Kbps channels respectively. GAO's G.723.1 software has switchable transmission rates over 5.3 or 6.3 Kbps channels providing 4 kHz of quality speech bandwidth in conformance with the ITU-T standard. The higher bit rate has greater quality. The lower bit rate gives good quality and provides system designers with additional flexibility.

The ITU-T G.723.1 standard specifies a coded representation that can be used for compressing the speech or other audio signal component of multimedia services at a very low bit rate as part of the overall H.324 family of standards. This coder was optimized to represent high quality speech at the above rates using a limited amount of complexity. It encodes speech or other audio signals in frames using linear predictive analysis-by-synthesis coding. The

excitation signal for the high rate coder is Multipulse Maximum Likelihood Quantization (MP-MLQ) and for the low rate coder is Algebraic-Code-Excited Linear-Prediction (ACELP).

TOS (Type of Service)

TOS has 8 bits reserved to the service type in the IP datagram. 0-2 means precedence. 6-7 are unused. 3-5 means D (requests low delay), T (requests high throughput), R (requests high reliability), respectively.

T.38 Standard Fax Protocol

T.38 is an ITU-T Recommendation. T.38 describes the technical features necessary to transfer facsimile documents in real-time between two standard Group 3 facsimile terminals over the Internet or other networks using IP protocols. The Recommendation allows the use of either TCP or UDP depending on the service environment.

Redundancy Frame

Redundancy frame is used to ensure more reliable fax transmissions than that of RTP voice transmission.

V.21

V.21 is an ITU-T recommendation for full-duplex communication between two analogue dial-up modems using audio frequency-shift keying modulation at 300 bauds to carry digital data at 300 bit/s. If fax machine doesn't get any fax signal, the gateway can detect any fax signal throughV.21.

NSF（Nonstandard facilities）

Non-standard fax facilities are those whose operation features are not defined by ITU. Some of those features are encoded in FIF but their encoding method was not defined.

Request Line

Request Line and Status Line are one of the RFC 822[9] defined information formats to carry out real time transmission task. Request line uses a method field right after the URI and protocol version fields to indicate request type, and ends with CRLF (Carriage Return or Line Feed). The request can be divided by spaces, but not Carriage Returns or Line Feeds can be used unless when reaching the end of the request.

Via

Via indicates path of route request. Via portion explains origin, route request time, route destination, and port. For example, R 128.200.10.0/24 [120/1] via 128.200.1.1, 00:00:17, ethernet0/0.

Border Agent

Border Agent includes Sign Proxy and Media Proxy.

RC4 Algorithm

The RC4 encryption algorithm is stream cipher, which can use variable length keys. The algorithm was developed by Ron Rivest, for RSA Data security. Analysis shows that the period of the cipher is overwhelmingly likely to be greater than 10100. Eight to sixteen machine operations are required per output byte, and the cipher can be expected to run very quickly in software. Independent analysts have scrutinized the algorithm and it is considered secure.